# RECORDING ENGINEER / PRODUCER

$4.00
December 1985
Volume 16 — Number 6

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

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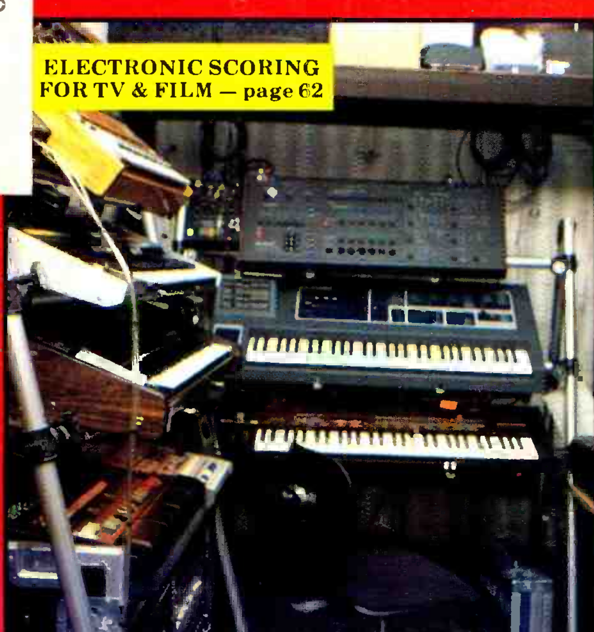

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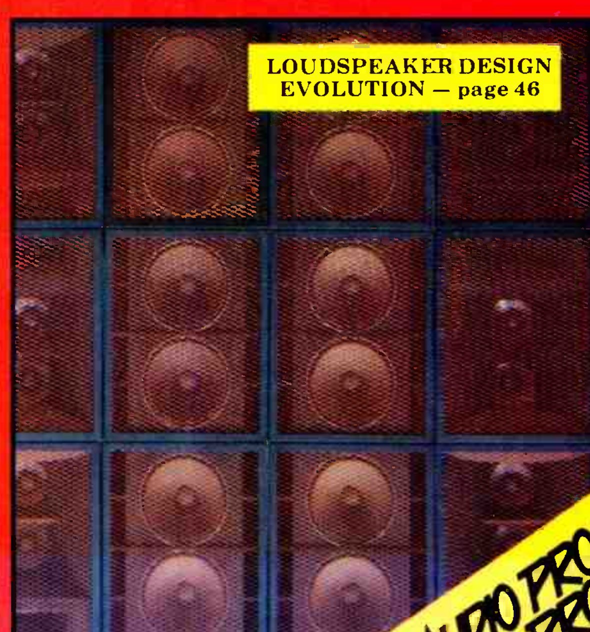

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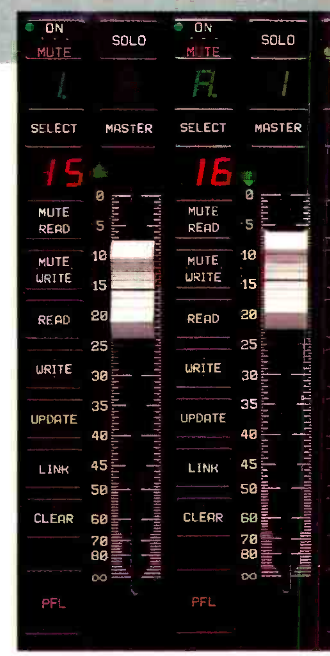
# SUPERSTAR

by quad eight




Advanced technology and unparalleled flexibility come together in the SUPERSTAR music recording console. Development of this console centered around the dual requirement of truly high definition sound and low noise, so critical for digital recording.

No other single console offers the combination of superior sound and flexibility in size and layout at such an affordable price. Field expandable, the SUPERSTAR provides ergonomical positioning of the console modules, allowing you to satisfy your own configuration needs. High resolution meters, central bus assignment, Intelligent Digital Faders, and the most comprehensive automation system all add up to SUPERSTAR—your next console.

## MODULAR CONSOLE

The SUPERSTAR is a totally modular console using air frame design concepts for strength and rigidity. Individual frame sections are in groups of 8 modules, with plug in wiring for true field expandability. The modular overbridge accepts the new limiter/compressor/gate for use either in-line with the input module or as a peripheral.

60-segment LED bargraph meters use advanced circuitry for precise and stable indication, offering VU, Peak, VCA level, and Spectrum Analyzer displays switch selectable.

Plug-in interchangeable equalizers and preamplifiers in each I/O module give instant user selectability and allow the addition of new technology at any time. Each module is of dual-purpose in-line design with line trim, equalizer, filter, 8 echo/cue sends, and fader switchable into the monitor/mixdown or main channel. Monitor/mixdown can be assigned to two independent stereo output busses for added versatility.

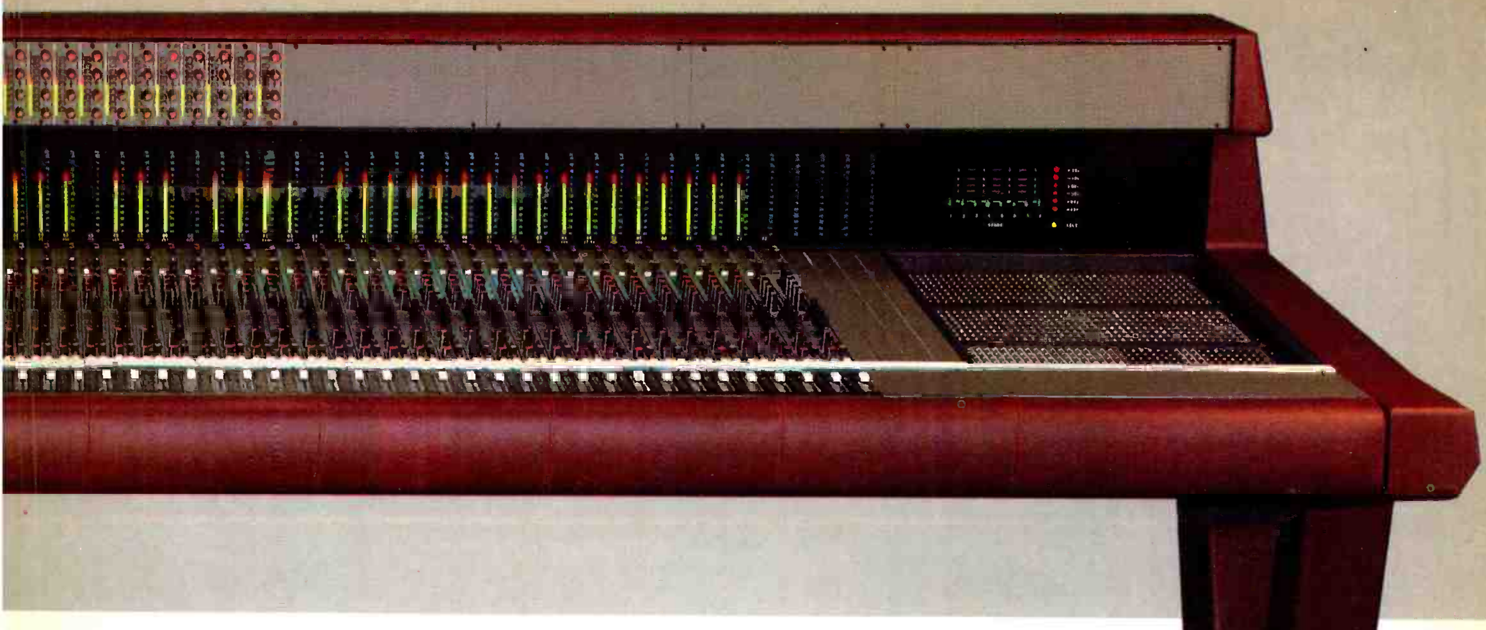
## CENTRAL ASSIGNMENT

This electronic output assignment cross-point switching system assures fast and reliable connections from the console to your tape machines with full routing or mixing capability. 64 output busses are assigned from each input module by a central touch control plasma display panel controlling up to a 96 by 64 electronic switching matrix. Completely software driven, the panel allows instant selection and display of the bus assignment with 10 presets in local memory. Optional unlimited storage to disk is provided. Easy to use, the system prompts for bus assignments and provides help through informative menu displays.

The building block matrix system consists of 16 by 16 switching cards bussed to 16-output summing cards. Logic controlled monolithic switching elements use zero volt current switching for extremely low distortion and feed through.

## THE NEXT GENERATION

The introduction of the SUPERSTAR signals a new era in professional sound control. With more and more studio facilities acquiring digital multitrack recording capabilities up to 64-track, larger sophisticated console systems with transparent sound performance are necessary. Digital signal processing (DSP) is neither economically feasible nor technologically advantageous today. A new generation analog console with advanced digital control is required to bridge the gap between the DSP consoles of the 1990s and the currently marketed analog consoles of the 1970s. The SUPERSTAR is such a console system. See it before you decide.



# COMPUMIX IV

*"A giant advance in automation accuracy and performance."*


The Fourth Generation Console Automation System is here. Compumix IV advances dynamic automation technology far beyond the capabilities of other systems, to a level of sophistication and accuracy demanded by tomorrow's digital recording techniques.

The FORTH realtime software running in a 32-bit 68000 computer provides 4 simultaneous mixes on-line as well as write command recall accuracy of 1/10 frame. SMPTE time code driven, Compumix IV stores *every frame* (not only changes) making it possible to perform editing functions on-line. This requires an 80 Mbyte hard disk storage system designed for fast access in both read and write modes.

Compumix IV is designed to control up to 256 IDF fader functions in realtime through easy to operate touch-sensitive plasma control panels. An optional Graphics Display System is available.

Nothing can touch it—except you.

## INTELLIGENT DIGITAL FADER

The IDF is a microprocessor-based module that utilizes the most advanced technology available. The super smooth fader is a 10-bit digital encoder that supplies 0.25 dB resolution and 119 dB of dynamic range. The grouping functions are the most extensive ever supplied in a music recording console. 16 groups are assignable with 4 levels of operation: slave, group master, submaster, and grand master.

Up to 256 IDFs run independently through a revolutionary "back door control bus" without the need for external computer automation. Realtime display of dB level, groupings, status, fader position and mutes are available at all times. 9 membrane switches allow for selection of up to 160 software defined functions.

From VCA to servo level control, the IDF is the next generation in fader technology for analog and digital console systems.

*For more information, please call or write*

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**For additional information circle #101**

# News Letters Views

## PUTTING PEOPLE FIRST

*from:* **Scott Spain**
**independent engineer**
**Granite Falls, WA**

I have been actively supporting myself and my family of five for the past 13 years by working in the music business as an engineer. During that time, I've worked with major artists, as well as neophites looking for their golden break.

Throughout my 13-year career, I have tried to stay out of the fast-lane lifestyle of Southern California, of which I'm a native, and have tried to live a more laid-back life among the trees and streams. I choose a lifestyle for my daughters over career advancement for myself and, within my world of small-town recording, have discovered the most important process of record building: the creative people and their inner sensitivities.

God bless the engineer who can take a group of sensitive people in a sterile environment — microphones don't applaud — and make them feel at home, helping them to achieve maximum creative output while spending sometimes $2.00 per minute.

All too often while perusing interviews with people who have been in the right place at the right time — some not even in the business five years, yet enjoying great success — I read over and over about microphone techniques, effects used, and personal musical tastes and efforts. But little emphasis seems to be put upon the art of dealing with *people*.

Having dealt with "the Stars" and the beginners, I have found the Stars to be, in general, very knowledgeable about the recording process, and how to go about getting what they want, which makes my job fun. The *real* trick is dealing effectively with some one young and frightened by a bad previous experience.

In such cases, we get clients who enter into the studio feeling that they are going to have to hold the engineer's hand all the way through the sessions just to get what they're looking for. Nine times out of 10, however, these people haven't the slightest idea of how to go about getting the sound they want, and are afraid to admit that to themselves, and especially to the engineer.

Getting that person, or group of people, to relax and have undying confidence in the engineer is, in my opinion, the real *art* of being a recording engineer. Personally, I would like to see more space in the trades devoted to this art form; tricks in dealing with the sensitive people in our world would benefit the long-termers, as well as the new kids on the block.

My mentor, Don Sciarrotta, is in my opinion, one of the best in dealing with people. Donny enjoyed the reputation of being one of the "Great Ears" of his time. After spending thousands of hours behind Donny, I found the greatest asset to that Sciarrotta sound to be his personality and method of getting the people on the other side of the glass to perform at their maximum potential.

The best piece of advice I've received thus far in my career came from Donny. "Scotti," he once said, "Never forget that *every* client is your most important client." I've taken his advice to heart in a way that is now part of my life. I can honestly say that it has been, and still is, my greatest asset as an engineer.

I'll take a wild guess by saying that the majority of people that read the trade journals are at least semi-educated in microphones, tape machines, and processing equipment, and are aware that the way to explore is by doing, and *not* by reading about doing. Please don't take my words as those of one putting down the trade magazines. I find the information they publish to be invaluable when it comes to inspiring new ideas for microphone placement and electronic trickery. But if for whatever reason, the person on the other side of the microphone is uninspired, all the technical knowledge in the world is useless.

Our job is to capture their emotions and yes, the tape can tell (brutally so) whether your artist is inspired or not.

It becomes your world if it's *your* finger on the talkback! You're responsible for bringing the artist into your world, and making it part of theirs. Getting the artist to leave the outside world behind — and simultaneously accepting responsibility for keeping the outside world away from the artists until such a time that it will not affect them, and consequently affecting the sound on tape — is an awesome responsibility, and I would like to read more on how others deal with it.

I love this business; for me it's a business of people — sensitive, creative people in quest of capturing a part of their souls to share with others. I hope to see more in the trades concerning the art of dealing with these special folks.

Keep up the good work, ya'll. Our business needs magazines like *R-e/p*. ■■■

## SERVO-DRIVEN SPEAKERS

*from:* **Thomas J. Danley**
**Intersonics, Inc.**
**Northbrook, IL**

Regarding the recent article on the Intersonics Servo-Drive Louspeaker, which appeared on the October issue of *R-e/p*, a number of readers have indicated that the calculated sensitivity values for the SDL system seem too low to be doing what users claim. Those readers that used the correct formula had accurate sensitivities of the order of 93 dB 1W/m. However, the only SPL figure given in the article was the output at 22 Hz (116 dB at 32 VRMS), which is below the normal operating band, and was intended to show its performance as an extended-range, high-output subwoofer. As we had approached this field of acoustics from a rather unique position, we elected not to publish sensitivity figures for our systems. For low-frequency use, 1-watt sensitivity is not that good an indicator of how well a given driver will work as a high-power subwoofer.

Voice-coil drivers exhibit significant power compression, frequently losing in excess of 50% of its 1-watt (cold voice-coil) efficiency when operated near the full power rating (hot voice-coil). Also, for low frequencies, many drivers reach their linear excursion limit well below the electrical power limit. Under these conditions, increasing the power further does continue to raise the output SPL, but at the cost of double-digit harmonic distortion and abbreviated low-frequency output.

Some people mistake these excursion related upper-frequency distortion products as "tight low-end," but the "tightness" is usually level-dependent and becomes much less pronounced or even "hi-fi" at lower power levels.

While refining our designs, we realized that we had something completely different. The servo motor is free of position nonlinearity, and capable of handling large transients with low distortion. Excursion capacity (in-band) extends to greater than +4 dB over the continous-rated power. Because of the large amount of wire in the SDL motor, the thermal time constant is many times longer than the typical woofer voice coil. This fact, in conjunction with the internal protection circuit and the nature of musical power demands, allows most users to use two or three times the rated continuous power*.

For these reasons, our specification sheets indicate the frequency response and SPL measured at full power, which we feel is closer to real life than 1-watt.

*. . . continued on page 11 —*

*Amplifier and speaker-cable selection is more critical with SDLs than conventional speakers. An inappropriate amplifier on voice-coil drivers may not be "happy" driving the SDL, and the SDL may only deliver a fraction of its capacity. We are able to recommend several appropriate amplifiers quite highly, however. ■■■

# Eventide's Newest Harmonizer®. Bursts Upon The Audio Scene


## The H969 is coming through to deliver . . .

### The Cleanest Audio

The H969's new ProPitch™ digital electronic-splicing algorithm gives you the cleanest, most glitch-free pitch change ever. Deglitching is active over a wider bandwidth, too — a full octave wider. And Eventide has employed 16 bit linear PCM circuitry in the H969 for the first time in a Harmonizer, for superb audio performance in all modes.

### The Easiest To Use

We know that you don't always think in terms of "pitch ratios". Sometimes, you simply want to go up a major third, or down a fifth. So to make things easy, the H969 gives you twelve "instant" pitch change presets. Setting a precise major third, minor third, fifth, seventh or octave of pitch change is a cinch — just push one button. Each interval can be selected as a sharp (increase pitch) or flat (decrease pitch). There are also instant presets for sharp and flat micro-pitch change, for vocal doubling and effects.

You choose from two ways to select pitch ratios and delay times on the H969 — positional and auto incremental. Individual coarse and fine adjust controls make it a snap to get exactly the pitch ratio you want. And once you choose your settings, the digits are rock-stable. Unless of course, you ask the H969 to automatically vary the pitch ratio — up or down, at your choice of speed.

To make the H969 as easy to use in live performance as it is in the studio, we've included a front panel preamplified input, in addition to the usual XLR-type studio level input. Just plug in your instrument. There's a companion front panel output jack, too. The H969 also has remote line in/out switching capability, plus remote pitch ratio/delay time set provisions. A keyboard can also be accommodated.

### The Widest Delay Range . . . And More

With the H969, you get much more delay than we've ever put into a Harmonizer before — 1.5 seconds at full bandwidth (40Hz - 15kHz ± 1dB). Need even more? Just hit the "double mode" button and you can extend delay range to over 3 seconds, with 8kHz bandwidth. For added convenience, you can choose and save any five delay times for instant recall. Delay time and pitch ratio are each displayed on an independent readout.

### The Broadest Array Of Effective Features

The H969's full 1.5 / 3+ second digital memory is available in Infinite Repeat as well as Reverse Audio modes, dramatically increasing the versatility and usefulness of these effects. You can also vary the length of the reversed or repeated audio segment after capture. Pitch change, Flange and Doppler effects can be used in tandem with Reverse and Repeat modes.

Flanging on the H969 Harmonizer offers unlimited options. Flange sweep rate can be varied over a very wide range, or you can sweep manually. You can freeze the flange sweep at any point you select, and you can preset the point at which the flange sweep begins. We've also added a new Doppler mode.

### The Best Harmonizer Ever

The H969 is an addition to Eventide's full line of Harmonizer special effects units. Our industry standard H949 is still going strong. The H969 ProPitch Harmonizer maintains Eventide's leadership position. For your most demanding applications, the H969 represents the state-of-the-art in pitch change technology. Hear it at your Eventide dealer soon.

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For additional information circle #103

# ess Recording Studio

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If you're interested in relaxing in your home or studio 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 of three (not sold separately). Complete printed documentation is also available for $200 for each set of manuals.



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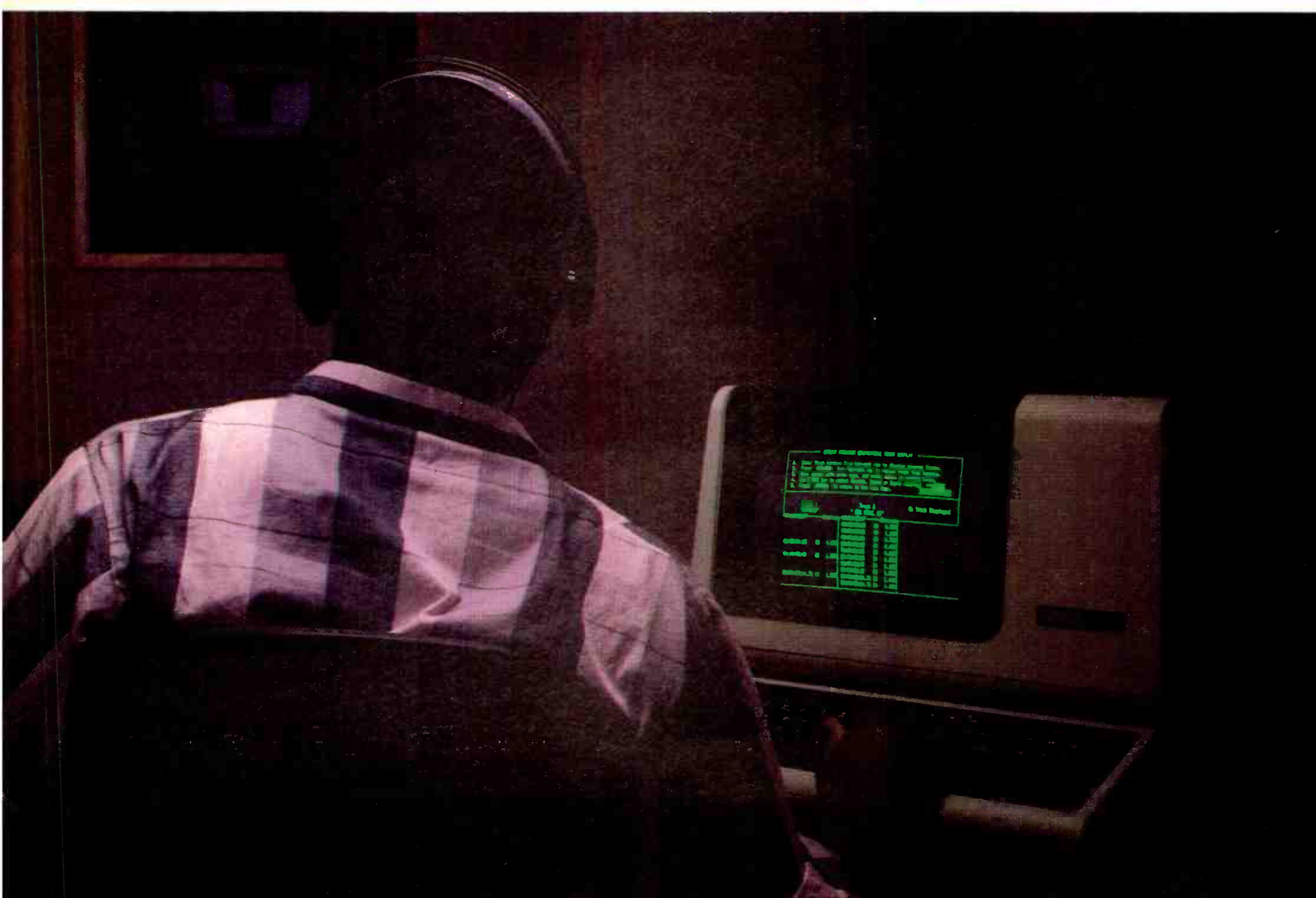

**For additional information circle #104**

*Synclavier operator captures continuous live vocal overdub*

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# LETTERS

## SERVO-DRIVEN SPEAKERS

— *continued from page 6 . . .*

As a point of reference, I have calculated the "1-watt" sensitivity based on our 32-volt (300-watt), one-half freefield measurements:

TPL-3: of the order of 103 dB at 1 W/1m; 28 to 100 Hz.

TPL-2: of the order of 102 dB at 1 W/1m; 40 to 100 Hz.

As these systems are often used in groups, which increases the horns effectiveness, improving sensitivity and frequency response, I have included those figures as well:

TPL-3: of the order of 100 dB 1 W/1m; 25 to 100 Hz.

TPL-2: of the order of 109 dB 1 W/1m; 35 to 100 Hz.

Four close-coupled TPL-3s, which have a combined mouth area of 45 by 90 inches, can deliver, with the proper amplifiers, low distortion musical transients, from 25 to 100 Hz in excess of 135 dB at 10 feet (1-meter measurements were *too* close for comfort!) ■■■

## SANKEN CU-41 MICROPHONE REVIEW

*from:* **Masao Konomi**
**Pan Communications, Inc.,**
**Exclusive Export Agent of**
**Sanken Microphones**
**Tokyo, Japan**

Regarding the review article by Professor Lowell Cross, University of Iowa School of Music, published in the October issue of *R-e/p*, we would like to make a few comments.

We were very much impressed with Professor Cross' obvious expertise in his field and with his fairness and balance in judgement. At the same time, however, we would like to suggest that he is perhaps entrapped in the conventional way of evaluating microphones; that is, he is relying on his subjective judgement as to the sound quality microphones are delivering to his ears. For example, he says, "To continue the comparison of the CU-41 and the TLM170, one could say that the latter has a slightly more 'velvety' quality in the midrange and highs."

We would like to point out to your readers that the Sanken CU-41 is designed to deliver a totally uncolored, distortion-free sound. As such, it picks up original information or sound to the maximum extent possible. It could deliver a more velvety sound during the stage of signal processing, if a sound engineer wanted to achieve that effect; in recording, the CU-41 picks up enough elements to make the sound "more velvety." Sanken designers believe that in the age of digital technologies, it is most important for a transducer such as a microphone to pick up sound information as precisely (that is as "uncolored") as possible. This is because once the original sound information is lost at the microphone end, it is impossible to recover at any subsequent stage of recording.

Any coloration or modification, which we believe is the area for sound engineers to contribute creatively to record or music production, should be conducted at a stage after the microphone.

Sanken engineers believe that the time has come for sound engineers to be freed from the inherent distortion of microphones, which often in a conventional way characterizes the sound quality of the microphones as devices delivering a "full," or "rich" or "warm" sound. With CU-41, sound and recording engineers can enhance their room for creativity by not being disturbed by the texture or distortion of the sound of th microphone in use.

In this connection, as Professor Cross pointed out, the CU-41 has achieved distortion-free performance. In other words, the CU-41 has a very linear frequency response over the full CD audio-frequency range, very low self-generated noise, almost clippping-free dynamic range, uniform polar patterns over the audio-frequency range, and little proximity effect.

In view of the recent, sudden acceptance of CDs in the market, this rather new way of evaluating microphones has become very important. There is a growing awareness among audiophiles in Japan that the quality of microphones used for CD recordings has become critical, since such quality can be heard. As a result, there is an increasing risk of recording music by a mediocre micro-

## LETTERS

**SANKEN CU-41 REVIEW**
— continued from page 11 . . .

phone in making a Compact Disc. After producers and sound engineers have made a big investment in signal processing and digital equipment, it would be a total waste to ruin the quality of original sound by using mediocre microphones.

Sanken has been pleased to find that their above-mentioned original design philosophies have started being accepted among top-ranking sound engineers in the world. Recently, we were told that with the CU-41, microphone setting time has been greatly reduced, because the performance of the CU-41 is so predictable. ■■■

**Editor's Note:** Because the above letter arrived close to our copy deadline, we were unable to secure a reply from Professor Cross for inclusion in this issue; we plan to publish a suitable response in the February 1986 issue — *ML.*

# EXPOSING AUDIO MYTHOLOGY

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

*by John H. Roberts*

This month I have gone back to previous columns of mine to dredge up two topics that I didn't completely finish writing about the first time around: gold plating and companding noise reduction. In addition, Part II of this column will take a look at consumer digital audio from a slightly different point of view.

### What Does Gold Plating Sound Like?

As my recollection of freshman chemistry is a bit rusty (oxidized?), I will try not to get in too deeply invlolved.

Before the increase in gold prices, gold plating was widely used on switch contacts, IC sockets, and printed-circuit edge connectors. Although the price of gold has increased several-fold, it is still widely used, albeit more selectively. Gold is attractive for these applications, because it is among the most inert of all metals.

While freedom from oxidation is desirable for ornamental purposes, it is also useful electrically. Most metallic oxides are poor conductors of electricity, resulting in oxidized contacts that can suffer increased resistance, semiconduction, or even open circuits. While heavy currents can break through thin oxide layers, audio signals fall into the category of light current or "dry" switching and require a clean metal-to-metal contact for acceptable results.

What all this means for those of us on nickel-plated budgets is that contacts should be kept clean, and exercised at some interval. The process of plugging a cable into a jack will usually scrape through the oxide layer and restore intimate contact. *Do not* use steel wool to clean your contacts, because stray steel whiskers will cause more problems than cleaning the jacks with such as abrasive might solve. Several nonmetallic scouring pads sold at your local supermarket will do the job just fine.

While I don't expect to hear any audible difference between a good gold-plated connection and a good tin- or nickel-plated one, an oxidized contact can and will sound bad. If you are forced to work in a hostile environment (such as producing outdoor sound) consider regular connector maintenance — perhaps even going for the gold!

One final caution: While I expect the extremely high passivity of gold will preclude catastrophic result, I am not comfortable with the idea of mixing connector types. Combining dissimilar metals is how batteries and thermocouples are made. Personally, I don't desire either of those devices in my audio chain.

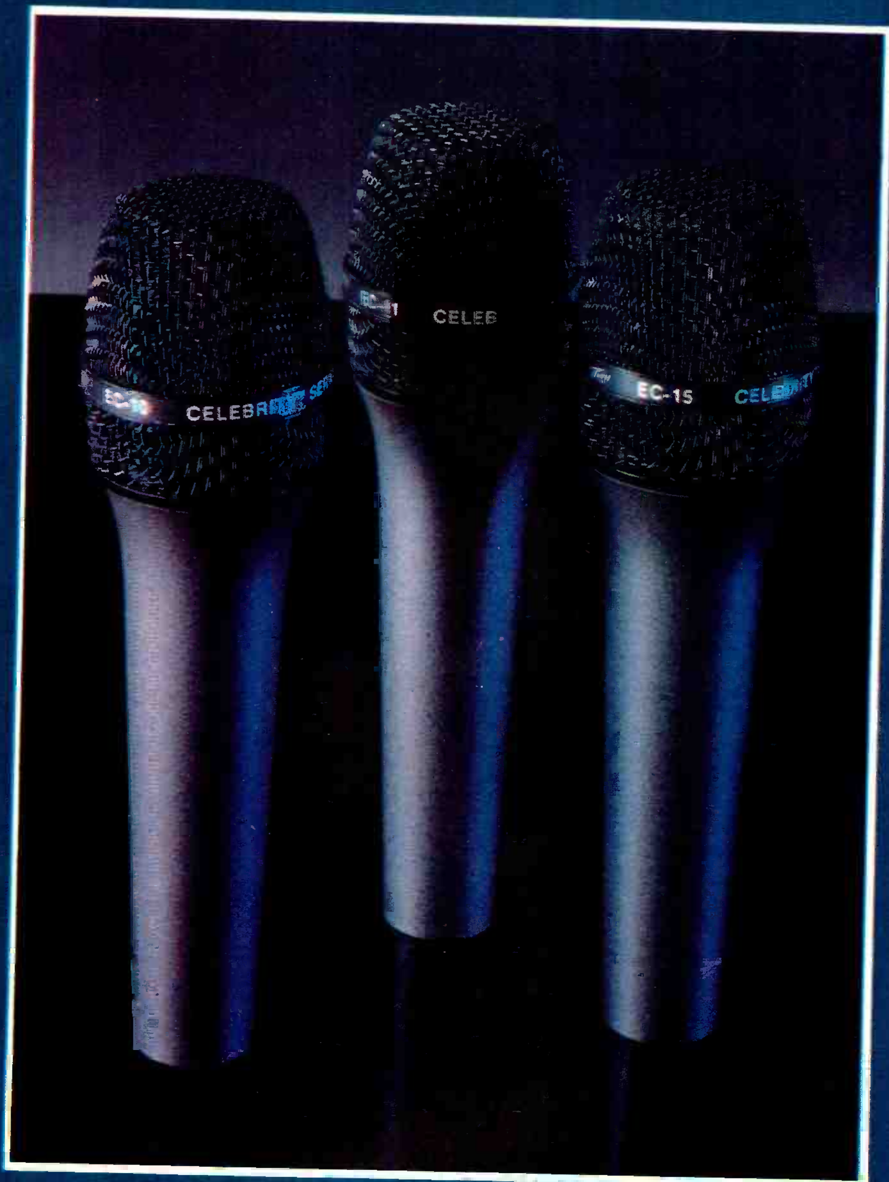
### No Free Lunch

Some time ago, I promised to delve into the trade-offs encountered within companding (compress to encode/expand to decode) noise-reduction systems. Compansion is used much more widely than you might first guess. Besides the ever-present Dolby, Telecom and dbx tape noise reduction found on most professional machines, companding is used widely on just about everything from film tracks to satellite feeds.

Applying compression to a signal before sending it over a noisy channel, then applying a complementary expansion at the receiving end, can dramatically reduce the noise contribution of that channel. The amount of benefit you receive is a direct function of the amount of compression/expansion that you apply. While promising the most noise reduction, an infinite compressor (AGC) wouldn't work, because the playback decoder must sense a changing level to

. . . continued on page 17 —

# AND NOW, FOR YOUR NEXT PERFORMANCE




## Celebrity™ Series Electret Condenser Microphones

The EC™ 10, EC™ 11 and EC™ 15 electret models are the latest addition to our popular Celebrity Series. Their condenser design offers enhanced transient response and reduced sensitivity to mechanically generated handling noise.

Cardioid patterns of the EC-10 and EC-11 are well-defined and smooth to provide superior rejection of unwanted off-axis signals. The EC-10 is designed to have a flat, natural sounding frequency response when used in close proximity to the sound source. The EC-11 has a wide, flat frequency response when used at a distance of six inches or more and a mild, controlled bass emphasis when used as a hand-held vocal microphone.

The EC-15 omnidirectional microphone has broad bandwidth and uniform response which recommend it highly in situations not requiring a unidirectional polar pattern, such as broadcasting and recording.

The Celebrity Series microphones from Peavey. Quality and performance at prices you'll applaud.

*Peavey PS™ 2B and PS™ 4AC Phantom Power Supplies are available as optional equipment for use with the Celebrity Series Electret Microphones when the mixer lacks integral phantom power.*

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For additional information circle #108

PEAVEY®


# Why should anyone else lis

A good monitor mix lets each member of the band hear exactly what he or she needs to hear. And that can be critical in helping any band play its best.

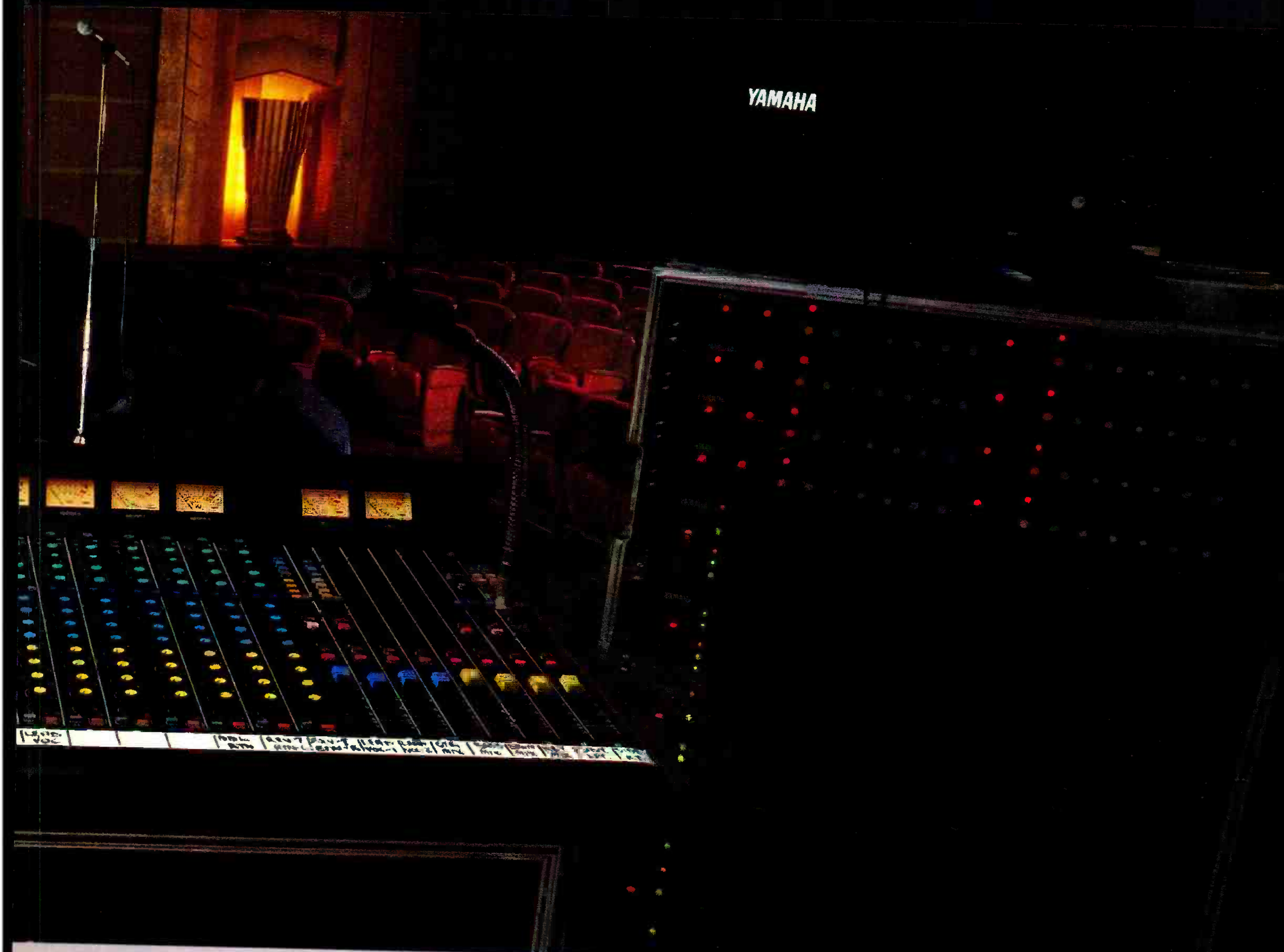
It's with this in mind that Yamaha designed the new MC monitor mixing consoles, the 16-channel MC1608M and the 24-channel MC2408M.

Both offer eight independent monitor mixes via eight outputs. And two auxiliary sends which can be used to patch in signal processors or tape recorders, as well as provide two additional mixes.

Each channel has a phase reverse switch, 20 dB pad, gain control, peak LED, three-band equalization with sweepable mid-frequency, two auxiliary send controls, eight rotary level controls, channel on/off switch, and channel cue switch.

The Input Channel Cue Priority System makes the monitor mix engineer's job a little easier. By pressing the cue switch on one or more input channels or auxiliary inputs, the master outputs being monitored are muted. So he can monitor only the selected input signal through headphones or speakers.

The MASTER section cue function permits the engineer to monitor an individual performer's monitor mix through headphones or speakers without affect-


YAMAHA

# DREAM MACHINE.

If mastering or post production is your business, our new MTR-20 is your dream machine. It's the only new ½" four-channel with 14" reels, to give you one hour recording at 15 ips. It also comes in ¼" center-track time code, ¼" two-channel, and ½" two-channel versions. And, the MTR-20 features Otari's exclusive Auto-Alignment for fully automatic calibration of record adjustments.

You can program the "20" to fit your personal style, with over 400 different set-ups. These range from controlled wind speed to a complete re-ordering of the transport control buttons—you can even make it left-handed if you want! You see, there are so many different applications for the MTR-20 that, rather than lock you into just *one* mode, we let you configure the machine to your specific needs.

Call your Otari dealer today, and wake up to the Dream Machine tomorrow. From Otari: The Technology You Can Trust. Otari Corporation, 2 Davis Dr., Belmont, CA 94002. (415) 592-8311 Telex 9103764890.

For additional information circle #110

OTARI®

— continued from page 13 . . .
compute its original gain.

The most popular compression ratio in use is 2:1, because it is also the cheapest to implement. Simply multiplying or dividing a signal by its own amplitude will result in the popular 2-dB input change equals 1-dB channel change, which, in turn, equals 2-dB output change [log(x squared) = 2 log(x)]. Higher ratios can be generated by cascading additional sections, and are sometimes used for satellite communications. Lower or non-integer compansion ratios require more sophisticated and therefore more expensive processing.

The basic consideration involved in selecting a compression ratio is how much noise reduction you need versus how much expansion of frequency-response errors you can tolerate. For simple signals, a 2:1 compander will be down 6 dB at the transmission channel's -3 dB point. For complex signals (like most audio applications) you can somewhat reduce this effect by band-pass filtering the level detector, but it will always be more sensitive if you use a higher compression ratio.

For applications attempting lesser amounts of noise reduction, a compression ratio near 1:1 would be very attractive, but prohibitively expensive. For that reason, systems such as CX (and Dolby) use the easier to implement 2:1 ratio, and restrict operation to only part of the full input's dynamic range.

Another major design problem while working with companding noise reduction systems is their inability to instantaneously change gain. Large, sudden transients will cause compressors (or the transmission medium) to momentarily saturate, while they wait for the gain control to catch up. Systems that are intentionally limited to lesser amounts of noise reduction will be less sensitive to this problem. While in theory, a compressor could be designed to have an arbitrarily fast responding time, waveform distortions, bandwidth limitations, and other errors in the transmission medium would make the audible cure sound worse than the problem when the signal is subjected to complimentary expansion.

The most popular design approach is to make the attack time fast enough so that you can't hear the overload (we are not very good at discerning brief bursts of distortion on the leading edge of signal attacks), yet concurrently slow enough to be unaffected by transmission related waveform distortions.

The final consideration when looking at companders concerns how they interact with the transmission channel's noise floor. Not unlike the banker who only wants to lend money to people who are already rich, companding works better when the transmission medium is quieter to begin with. On very noisy channels, you might hear the noise floor falling and rising (often called "breathing" or "pumping") concurrently with the signal. This noise modulation is most objectionable when the signal spectra does not mask the noise spectra (like a bass guitar over tape hiss). Several techniques exist that address this problem, ranging from the simple, but quite effective, application of high-frequency pre/de-emphasis, to splitting up the signal into several bands, then companding each one separately. This latter technique is effective, but also more costly.

Dedicated systems give the designer an opportunity to optimize for particular strengths and weaknesses of a particular noise-reduction system. The dbx Model 700, for example, wraps a compander around a digital Delta Modulator. Because the Deltamod channel will normally have excellent frequency response and signal fidelity, the attack times can be made quite fast.

Another interesting variation on dedicated companders is the noise reduction being used on the MTS Stereo Television system, also utilizing dbx noise reduction. In this case, a major design consideration was the basic channel noise during reception by an audience in located outlying areas. The solution was to design in a variable pre/de-emphasis that combined with the basic 2:1 compander for more high-frequency noise reduction than a fixed pre/de-emphasis could offer.

I'm not sure what advice to give about using noise reduction in the studio. While tape and system noise might be intolerable without the application of NR units, you may encounter a track that just doesn't work (like close miking that 150mm howitzer!). If you don't like the noise-reduced sound, don't use it on that track. Since the sound usually will be something infrequent and percussive, a noise gate might help; but then only if such a unit screws up less than noise reduction.

### The Real World

I have wondered why so many people are buying and singing the praises of those $199 Compact Disc players, while the "Experts" are still complaining about flaws in this, by definition, imperfect record/replay system?

As happens too often during discussions within this column, both groups are reasonably right. We have discussed the limitations of sampled audio systems in past columns. Whether you find the present systems flawed or not, that decision would depend upon how you define "acceptable performance."

This month, I'd rather not look at why few people still object to CDs but, conversely, why so many don't. It is no secret that the Compact Cassette was (is) a roaring success because of its convenience — despite the fact that decent sound quality didn't reach the consumer until years after its introduction. While not indestructible, the CD is suitably

. . . continued overleaf —

rugged and very convenient to use.

Were the CD's sound quality only equal to a good cassette, it would probably still be a viable product; but, as it sits, it is in serious competition with a tweaked out record player.

If you think about it, the real competition *isn't* between the high-end consumer systems. The differences in sound quality between the most- and least-expensive CD players (Yes Virginia, there *is* a difference) are fairly subtle. While you may hear the difference on a "big-bucks" (high-resolution) system, a typical consumer replay system will mask these detrimental differences. What the typical hi-fi system won't mask, however, is the deficiencies of a typical record player.

The differences between record-playing systems are much grosser. While the best turntable can sound transcendent, a lesser system is very much a lesser system. And all this is compounded by the fact that even the best system improperly set up will sound dismal. While the following speculation is just a guess, I am willing to predict that half or more of the records being played today are dirty and/or damaged, using dirty and/or damaged styli in improperly adjusted cartridge/tone arms (VTA, anti-skating, tracking force), terminated by a pre-amp that is neither flat nor provides an optimum load for the cartridge's reactances.

It should come as no surprise that consumers are blown away by the improvement in sound quality that CD offers. If history repeats itself — and I see no reason why it shouldn't — things in "CD Land" will only get better. How much record player (with pre-amp) can you get for under $200? (I expect to see CD players hitting even lower prices by the time this column appears in print.)

The implications for us in the professional end are several. First, keep your needle clean. More importantly, keep your tracks clean. In the good (not so) old days, a lot of noise and garbage got buried in a record's surface noise. Now even the cheapest CD-based system will deliver close to a 90 dB dynamic range. If you leave some hum on the bass track, or forget to shut down the vocal mike between verses, the grunge will be there for all to hear. As with other technical advances, the standards of acceptable sound quality just got ratcheted up another click. This higher standard will ultimately translate to better sound quality in other mediums as more people learn how good audio can sound, and thus, become more discriminating. I can hardly wait. ■■■

# THE VISUAL MUSIC SCENE

## Towards a Universal Audio/Video Language: Roger Bailey of Paltex describes his plans for a "How to . . . " series of videocassettes on producing Music Videos

*by Adrian Zarin*

Right from the start of the present Music Video era, the alliance between pop music and promotional videos has been an uneasy one. It was only the direst of economic situations that first compelled musicians, record producers and audio engineers to stake their collective future on a new and alien medium — a medium with its own technology, aesthetics and business procedures. On one level, the alliance achieved its goal brilliantly by helping pull the record industry out of its mid-Seventies slump. On the other hand, musical artists, producers and engineers *still* tend to have their reservations about video. From time to time, they'll turn away from the screen, look at one another with quizzical expressions and ask, "What have we gotten *into* here?"

But what if the music sector could sieze control of the means of Music Video production? What if the very same people who make records could master the occult mysteries of shooting and editing Music Videos? While record people could never replace film and video professionals, it would clearly help everyone involved if these record-makers understood the video-making process a little better.

Although this is by no means a new idea, according to Roger Bailey of Paltex, it *is* an idea whose time has arrived. Since 1979, Bailey and Paltex have been involved in developing computerized video-editing and post-production equipment specifically designed to simplify post-production for video professionals, as well as such non-specialists as corporate users and music-industry people. Having made such a strong showing in the broadcast and corporate markets, Paltex is now bidding for the Music Video industry. Following the lead of many high-tech manufacturers in the pro-audio and musical instrument fields, Paltex has decided to play an educational role as a part of this overall marketing effort. The company will be producing a series of "How to . . . " instructional videocassettes that will cover the production and post-production aspects of making a Music Video.

Bailey himself hails from a combined marketing and technical background. After taking a university degree in electronics, he went to work for EMI Television, Phillips Broadcast, and Ampex in Great Britain. During the mid-Seventies, he was in charge of U.S. marketing and sales for Britain's International Video

. . . continued on page 22 —

# Which of these microphones looks "flattest" to you?

The flat response microphone is the one in the center. It's the new 4004 studio microphone from Bruel & Kjaer. A significant contributor to its flat amplitude characteristic and uniform phase response is its small size.

All other factors being equal, the smallest microphone will have the best response, because its envelope causes the least obtrusion into the sound field. If you'd like to see curves that show how flat a professional microphone can be, request our literature. If you'd like to hear how flat a professional microphone can sound, call your B&K field applications engineer for a demonstration in your space.

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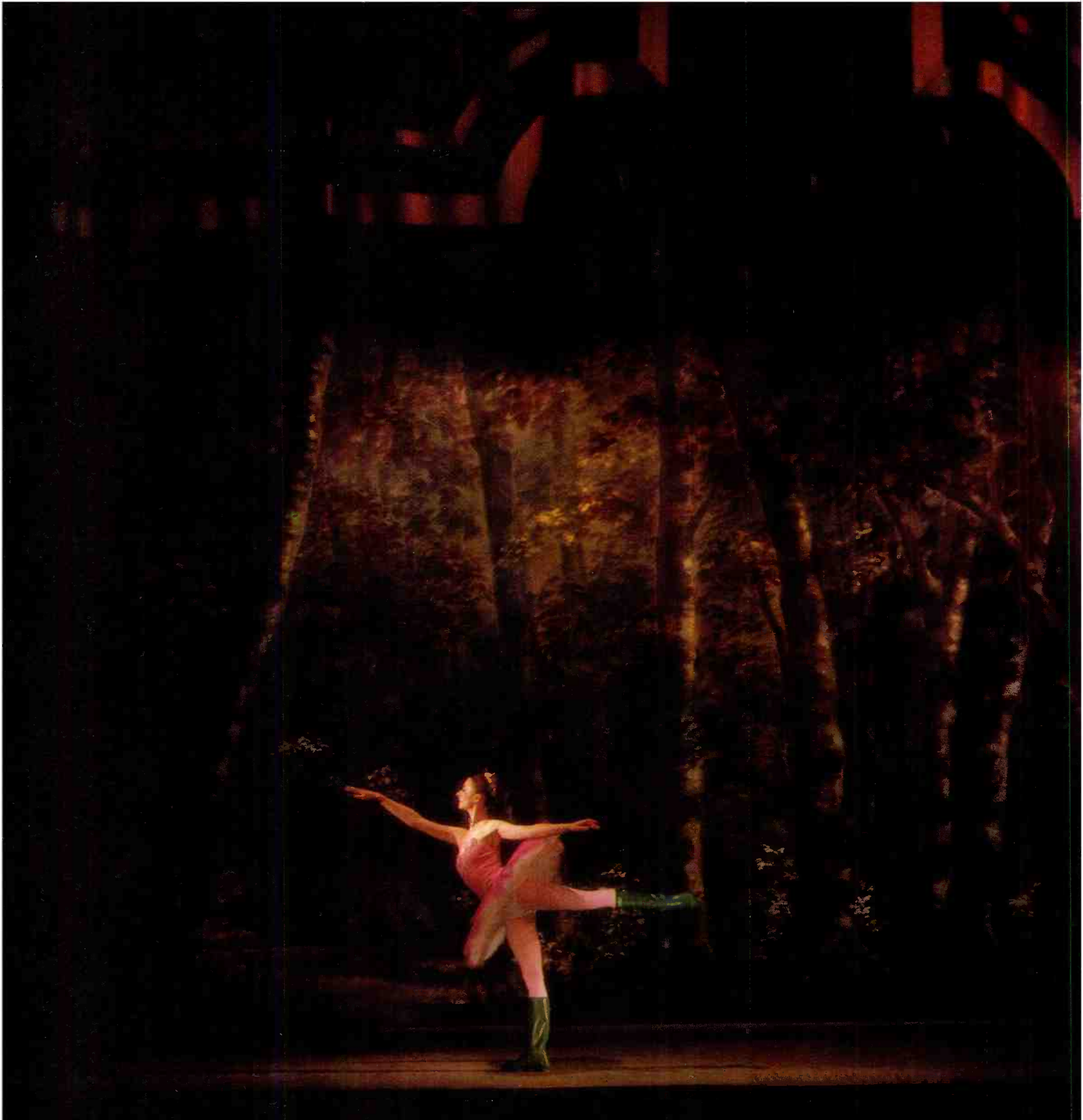

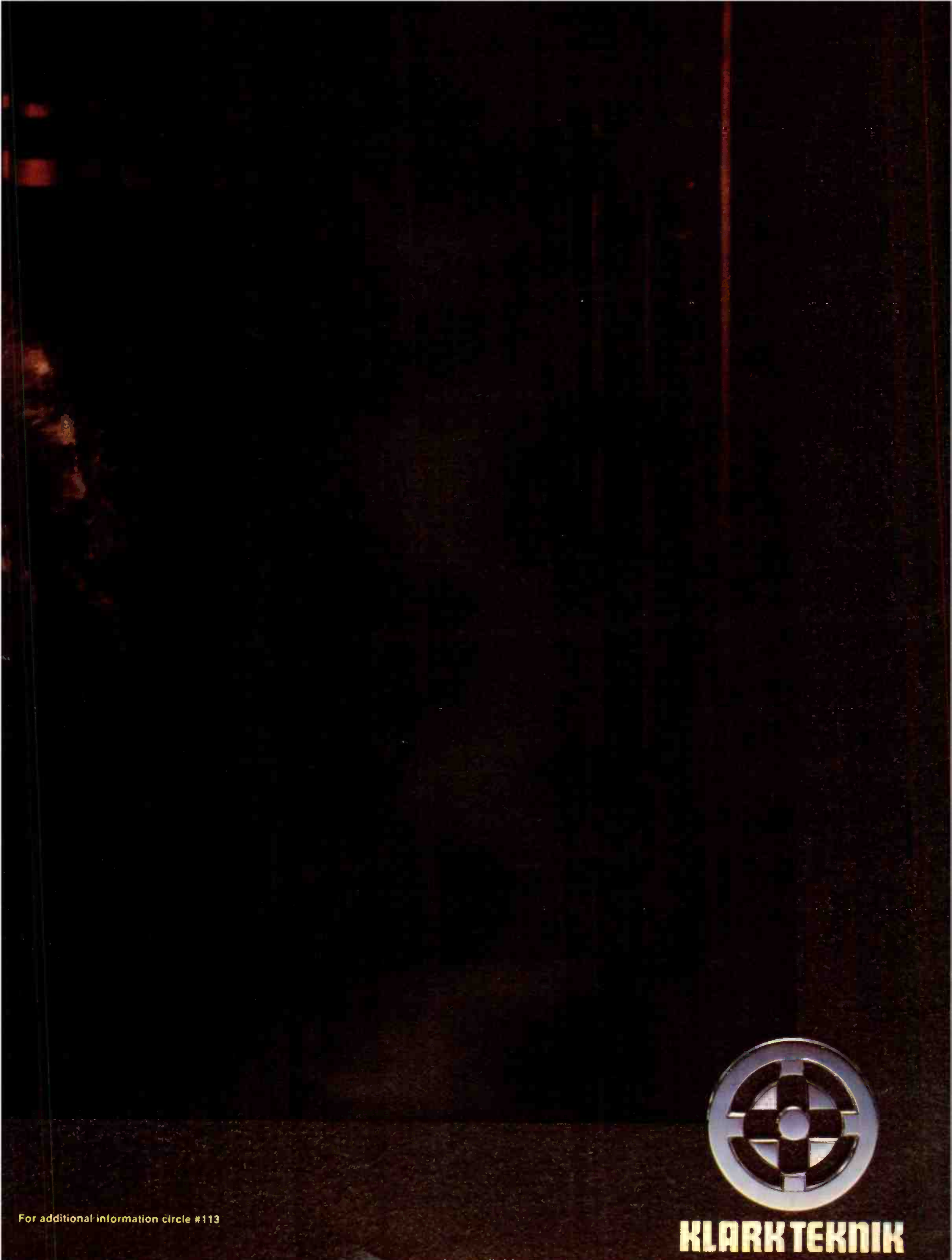
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KLARK TEKNIK

**VISUAL MUSIC** — *continued from page 18*

Corporation (IVC). In 1979, he started Paltex Editing and Production Systems, Ltd. with the initial intention of bringing computerized video-editing gear to European post-production markets. Acting at first as distributor for products manufactured by IVC and Datatron, Inc. of California, Paltex ultimately acquired both companies and became a formidable manufacturer of video equipment.

Paltex's late-Seventies launch just happened to coincide neatly with the early stirrings of the Music Video boom in the U.K. Indeed, the company's first London installation was for pioneering Music Video producer Jon Roseman and his seminal company, Jon Roseman Productions International. Ltd.

"This was in the days of two-inch [video] machines," Bailey recalls. "Jon had gotten together with a videotape editor, and they were looking for a way to do post production on cheaper equipment — namely, ¾-inch [U-Matic] equipment — and still have frame accuracy and all the other necessary features.

"They contacted me and said: 'We hear you're in the U.K. and we hear you're talking computer editing. We'd better get together.' The upshot was that Paltex put in a system for Jon. And so our introduction to London was straight through the music business."

The original premise of Paltex was that the European market had different expectations of computerized editing than did the American market. The U.S., says Bailey, was wholeheartedly prepared to embrace computerized editing in all its complexity. In fact, the more complex the equipment, the more job security for the person operating it. The Europeans, on the other hand, "didn't want to be buried in technol-

**STOP PRESS:**
**Neve Electronics Purchased by Siemens; Company Future Described as Secure Following Takeover**

At press time it was announced that Neve Electronics Holding Ltd.'s parent company, Energy Services & Electronics (ESE), now part of the Brammer Group, has reached an agreement with Siemens A.G. of Austria, and Siemens Ltd. of England, for Siemens to take control of the Neve subsidiary by the beginning of next year. As reported in the August issue of *R-e/p*, it had been expected that Neve Audio would be sold following the recent purchase of ESE by Brammer.

Commenting on the announcement, Laci Nester-Smith, Neve Audio's group managing director, said: "The removal of uncertainty about our future ownership will be welcomed by our customers and suppliers alike. The individual product and market positions of Siemens and Neve Audio complement each other. As a result, the world audio market will get a more comprehensive service from this combined group. Siemens is committed to the future development of Neve's technology in both the analog and digital fields, and significant R&D programs are planned."

ogy," Bailey remembers. "They see their jobs as part of the creative process. So the approach we took was to go with that direction. We said, 'Okay, we'll provide a box. And we'll talk about this thing *just* as a box. We'll bury that in your equipment area, and give you as simple a keyboard as we can. And don't worry that there's a computer there. Just ignore it!'

"We wrote the software so that it wouldn't keep calling attention to the computer's presence by constantly saying 'Are you sure? Do you really want to do that?' In a piece of high-tech equipment, you don't keep coming back and asking things like that. You safeguard certain functions, such as memory dumps, and that's all that's necessary."

While Paltex initially directed its efforts toward the European market, Bailey began to discover that there were many sectors of the American market, apart from the professional post-production industry, where Paltex's "invisible computer" approach would be welcome. "We are finding that many new people coming into the business are just as nervous about computers as a lot of other people," Bailey notes. In this category, he lists film-production companies that are just getting into video; people involved in in-house corporate communications facilities; and, of course, a broad spectrum of music industry professionals, from record execs to artists.

"We've had a lot of interest from the bigger pop groups," says Bailey, "where they've wondered whether they should add video facilities to their recording studios. Take an artist like Alan Parsons, to cite an instance I'm very familiar with. For many years, Alan has played around in edit suites. He has never sat down and edited a finished video product, but has enjoyed manipulating video, if you like. Someone liked that says: 'I can sit in my studio and put together my master tracks; I also want to start playing with video, because it must be a similar technology.' We're seeing that quite a lot now."

Even a musician or producer who hasn't "played around" in video edit suites before can quickly get the hang of a basic computerized video editing system, Bailey claims. "He'll be 'editing' in 10 to 15 minutes; he'll be into dissolves in half an hour. Now alright, if he gets into heavy edit decision list management and some of these clever pieces of video time compression and expansion, that will take a little longer. But he'll learn those pretty rapidly, too."

Granted, this kind of rudimentary grasp of post production will not yield a slick, 1985 piece of Music Video. But, in Bailey's view, it does have its use in the overall process of creating a music clip.

"Expanding on the concept of off-line, what I see a lot of these groups doing is 'off-off-line.' Before they really go into serious off-line, they can literally put their video together and see how it goes.

"As Music Video develops, people are

**Roger Bailey of Paltex**

getting away from the large four- or five-camera shoots and the use of iso reels — that was something that had come across from the film world. Now many people are often shooting true single-camera style. You may end up with many, many reels of videotape, and that calls for a lot of putting together in post production. So you're seeing more and more musicians beginning to get involved in that preliminary process of 'How do I start to link all of these segments together? Then, when they get to the edit suite, they're saving time — and, of course, money."

What Bailey calls "off-off-line" is really a video expansion of the idea behind a musician or producer's personal-use studio. As with many home studios, the idea is not to come up with a finished product, but simply to provide a sketch or demo that can be used to guide the making of the finished product. It creates an opportunity to experiment without having to pay for the studio time; and it lets the artist put more of his or her own personal stamp on the finished product. But, unlike a modest home recording set-up, an entry-level "off-off-line" video editing suite can run into quite a bit of money; Bailey puts the figure at about $40,000.

"You can do something for less," he explains, "but then you're restricted to just cuts, which means you're going to end up frustrated. The moment you say, 'I have to dissolve here,' you need other digital boxes to do that; so you're talking $40,000 to $50,000. But, of course, if you were making a video album, for example, you could spend that type of money renting a facility just to find out that a particular idea doesn't work after all."

### Record Company Interest

Apart from musicians and record producers, says Bailey, record companies themselves are also starting to express an interest in having their own video facilities. Last year, he reports, Paltex built a large video facility for Chrysalis Records U.K. and Music & Management (MAM), two companies that have since merged.

"The interesting thing here," he comments, "will be to see if they put in their own in-house directors. Of course, it's too early to say, but I expect not; I suspect that they'll continue to go to outside TV directors. Now, however, the situation will be: 'Yes, we have this number to be directed, but you'll use our facilities to do it.' Which, in the end, will still save them a lot of money." *continued overleaf —*

Installations such as the Chrysalis/-MAM facility have convinced Bailey that the music industry is ready to start playing a hands-on role in the making of Music Videos. Making easy-to-use hardware and software available, however, is only part of Bailey's plan. The other part is Paltex's projected series of "How To . . ." videocassettes on producing Music Videos.

The project grew out of two specific considerations, says Bailey. "First of all, there's a lot of nervousness in the recording industry about Music Video. Record companies and record producers recognize that, if their record is going to be a success, a Music Video must be made. But they really don't know how one is made. So a lot of people are wondering if they're getting ripped off. They're uncomfortable about passing all responsibility over to a television director and saying: 'Our life rests in this man's hands.'

"Secondly, editor friends of mine would always comment that they'd just spend 48 hours in the edit suite working on such and such a project; and if only the guys had really known what it was all about they wouldn't have needed to spend so much time and money."

"There just seems to be a complete lack of knowledge as to how a Music Video is made; there aren't many good textbooks on video production and post production. Something as visual as this really can't be taught in a textbook. So our concept was: let's make a Music Video; but then let's also tape the making of the video to show how it's done. It's quite a complex program, really. 'Shooting a shoot,' so to speak."

### "How-To Videocassettes"

According to Bailey, Paltex is planning to produce three, 40-minute cassettes covering the basics of Music Video production and post production. He's drafted video editor Greg Griffiths and director John House (Air Supply, Pat Benatar, Hall & Oates, Yes, Mick Fleetwood) as consultants. He also plans to involve a prominent lighting director (as yet unselected). The basic idea is to draw on the resources of professional that routinely have to explain video to laymen colleagues and clients as part of their everyday jobs.

"Over the years," Bailey elaborates, "Greg has continually been confronted with the situation of saying, 'Tell me in your own words what it is you want to put across,' and then translating that into the appropriate techniques. So we're using Greg for the story line, if you like.

"We also have John's input. Again, here's a director confronted with making videos and having to explain things on a day-to-day basis. We'll get the same kind of input from our lighting person as well. It sounds terribly as though it's going to be done by committee, but I think it almost has to be done that way. You have to be careful that you don't get any one particular bias in a project like this."

What makes a project such as Paltex's instructional tapes particularly timely right now is the fact that audio and video technologies clearly seem to be converging. Bailey cites computer editing and digital program-storage media as the twin technological bases upon which a true audio/video union will firmly stand.

"At the most recent broadcast and video shows," he remarks, "people were starting to appear with genuine audio editors which had been adapted from the way video editors work. So things are slowly coming together. And then, of course, we've got digital around the corner. When it arrives, we'll all basically be working within a PCM format of one type or another. At the present moment, there's some dispute over what

## TIME DOMAIN PROCESSORS AND SPECIAL EFFECTS UNITS

### Directory Omissions and Corrections

Since the compilation of this Directory, which appeared in the August 1985 issue of *R-e/p*, we have received details of the following delay and reverb units:

**LT SOUND**
**P.O. Box 338**
**Stone Mountain, GA 30086**
**Phone: (404) 493-1258**

**Echo Control Center**

**Inputs:** Four.
**Outputs:** Two.
**Effects Type(s):** Delay and reverb unit.
**Delay Ranges:** 0.3 milliseconds to 1,000 milliseconds.
**Dynamics Parameters:** Over 18 kHz frequency response; 0.6 seconds to 2.2 seconds of decay.
**Operational Controls:** EQ; decay; amount; chorus; flange; double; eight preset controls.
**Frequency Response (input/output):** Delay — 10 Hz to 20 kHz; 20 Hz to 18 kHz on reverb.
**Distortion (input/output):** 0.06% THD at 1 kHz.
**S/N Ratio (input/output):** Better than -94 dBv.
**Pro-User Price Range:** $995

**Model RV-2**

**Inputs:** Eight.
**Outputs:** Two.
**Effects Type(s):** Reverberation: acoustic chamber; two different plate sounds.
**Operational Controls:** Selectable reverb sound; decay; input/output level; seven LED meter display.
**Selected Standard Features:** The unit comes standard with half-track test tape — dry on one channel and return from EMT on the other —for comparison purposes.
**Frequency Response (input/output):** 20 Hz to 18 kHz.
**Distortion (input/output):** Less than 0.2% THD at resonance.
**S/N Ratio:** Better than -72 dBv; dynamic range 90 dB.
**Pro-User Price Range:** $895

**Model RC-1**

**Inputs:** Four.
**Outputs:** One.
**Effects Type(s):** Reverb system: mono, acoustic chamber, and plate.
**Operational Controls:** EQ; input level; decay.
**Selected Standard Features:** The unit comes standard with half-track-test tape — dry on one channel and return from EMT on the other — for comparison purposes.
**Frequency Response (input/output):** 20 Hz to 18 kHz.
**Distortion (input/output):** Less than 0.2% THD at resonance.
**S/N Ratio:** Better than -72 dBv; dynamic range 90 dB.
**Pro-User Price Range:** $595

the standard number if audio tracks should be once we do go to digital. The broadcasters feel that four tracks will be enough. But if you talk to the post-production people who are doing music, they'll need a minimum of eight tracks recorded with the video. Ideally, perhaps, there should be two formats: a 'post format' and a 'broadcast format.' [Bailey is referring here to the proposed SMPTE and EBU formats for digital videotape recording, which has provisions for a total of four PCM-encoded audio tracks on a digital VTR — Editor.]

"But there's no doubt that the technologies are merging. There was a time when, on the one hand, we were talking digits and computer control; and on the other hand, there was a musician who played his guitar, and everything in the world was analog. Now, that's all going away — particularly among the younger generation. Today's teenage computer buff will be tomorrow's TV director, TV producer, record producer, composer, etc. They'll all be coming from a common ground. As we get older, we're all being replaced by people who have no problems with computer technology."

### Blending Audio with Video

This much is clear. In the past, we've seen some of the most sympathetic, visual treatments of pop songs come from the directorial hand of the musicians and producers who created the songs in the first place. The video work of Thomas Dolby, DEVO's Gary Casale and Talking Headsman David Byrne are the examples that immediately spring to mind. And, at the time of this writing, a relatively new, highly musical video style is gaining popularity in the electronic/dance music genre. Tracks like "People Are People" by Depeche Mode and "19" by Paul Hardcastle have been effectively mated with montage-style clips that are kinetically cut to the beat of the music. Clips like these make explicit connections between present-day recording technology and present day video editing technology. A vocal line or riff that's been caught in a DDL loop (dance-mix style) will be paralleled by a video image looped in perfect sync with the audio. A quick hit from a Fairlight orchestra sample, for example, will be matched with a rapid cut to a shot of an actual orchestra — again, right on the beat. If you're looking for a visual rendering of the mental processes that take place when a person listens to music, clips like these come closer than narrative storylines, or even straight-forward performance clips.

Maybe it's no coincidence that the aforementioned Depeche Mode clip bears more that a superficial resemblance to director Sergei Eisenstein's 1925 classic *Potemkin*. After all, it was Eisenstein's bold editing style that spearheaded the theory that film should take music — rather than stage plays or narrative fiction — as its structural model. It's and idea that goes back to the infancy of filmmaking. But it's an idea that never caught on in the popular media — until now, that is.

As audio and video technologies continue to converge, perhaps the sense of uneasiness that still hovers over Music Video will come to an end. Perhaps we'll be a little closer to the true hybrid art form we've been promised all along.

□□□



# News

## MITSUBISHI, OTARI AND AEG ANNOUNCE PRODIGITAL RECORDING FORMAT

The PD (Professional Digital) format for recording digital audio on fixed-head transports, jointly developed by the three companies, claims advantages in sonic performance, reliability and flexibility over previous format attempts. PD-compatible tape machines will include 32 channels on one-inch tape at 30 ips, 16 channels on half-inch tape at 30 ips, and two channels on ¼-inch tape at 7½ or 15 ips, with both razor blade and electronic editing. The format agreement includes full tape, machine control as well as digital port compatibility between the different brands.

The PD format is currently being utilized in the Mitsubishi X-86 two-track and X-850 32-track machines. Otari is expected to show prototype PD transports at the European AES Convention in Montreaux in March, while AEG plans to develop a two-track PD machine by 1987; meanwhile the latter company will continue to market Mitsubishi products in certain European territories.

## UCLA OFFERING RECORDING ENGINEERING COURSE WITH GEORGE MASSENBURG

Massenburg will bring six students into his West L.A. studio for an intensive 12-session workshop on current techniques starting April 6. Those interested in participating in "Master Class in Recording Engineering with George Massenburg" must make an appointment for an admissions interview by March 14.

"This is an 'ear' course," says Massenburg, whose career includes production and engineering supervision credits on sessions with Phil Collins, Linda Ronstadt, Earth Wind and Fire, and Herbie Hancock. "We'll explore techniques for arriving at what students hear in their minds — in painting, this would be a visualization course."

The class will meet on five alternate Sundays at Massenburg's studio, The Complex, while a sixth class will meet at Ocean Way Recording, with engineer Alan Sides. On the six alternating Sundays, students will break into teams of two each for in-studio lab sessions at The Complex to work on individual assignments, with Massenburg available for consultations and feedback.

. . . continued overleaf —

# and this:

The advent of stereo video introduced a new level of audio post-production requirements. SSL responded with the SL 6000 E Series. This provides the same high standard of audio quality and signal processing flexibility as our 4000, adding a unique matrix to simplify the creation of separate stereo music, efx and dialogue mixes. The music video producers told us they needed to move projects freely between recording studios and post-production suites. We listened, and made both systems totally data-compatible.

The industry's acceptance was overwhelming. As a result, it is now possible to carry complete Total Recall ™ console setups, edit and cue lists, and dynamic mixing data between more than 250 leading music studios, post-production houses and broadcast facilities in 72 cities and 24 countries. In fact, this minor miracle happens every day. And to keep it happening around the clock, we've set up service offices and technical training around the world. Because we believe that your business deserves responsiveness, not promises.

**Solid State Logic**

Fee for this course, which is said to include many hours of studio time, will be $895. For more details call the UCLA Department of The Arts at: (213) 825-9064.

### LIGHTNING STUDIOS USES NEW REVECTORIZATION™ AUDIO PROCESSING TECHNIQUE FOR RE-RELEASE OF *THE WIZARD OF OZ*

Developed by Stephen St. Croix, president of Lightning Studios, a division of Marshall Electronic, Revectorization is a proprietary process that is said to enable audio noise and distortion to be removed from optical and magnetic soundtracks, plus similar media. In a related process, the now improved audio material is perspective matched to the accompanying visuals, and the "correct ambience and imaging constructed to provide a very realistic stereophonic soundfield from the original [usually mono] soundtrack," St. Croix explains. The process was used most recently to prepare an enhanced soundtrack for MGM's *The Wizard of Oz*, which was re-released on stereo videocassette in early November.

According to St. Croix, who has spent the last four years developing and perfecting the process, "It is possible to improve the dynamic range of a film that is only 40 to 50 dB, to as much as 70 to 90 dB, reduce distortion by as much as 10 times, and more than double its frequency bandwidth. The resultant stereo ambience and 'spaces' that the system provides are faithfully reconstucted to add impact to the visual images.

"The Revectorization process senses from the original audio the presence of obtrusive noise and distortion, and then eliminates these unwanted artifacts. The end result is a stereo soundtrack that draws a 'sonic picture' in the viewer's mind, and which adds a whole new dimension to their enjoyment of the picture."

For further details of the Revectorization process, contact Marshall Electronic, Lightning Division, on (301) 484-2220

### BASF CITES SUCCESS OF CHROME TAPE MASTER 920 AS KEY FACTOR IN DUPLICATORS' SWITCH TO CHROME

Since the introduction of its Formulation 920 Chrome Loop Master tape at the 1982 AES show, BASF has seen a "general overall improvement" in prerecorded cassettes available to the U.S. consumer according to Terry O'Kelley, national professioanl products sales manager for BASF Systems Corporation.

"In three years, the quality of prerecorded cassettes has improved to a measurable degree," O'Kelly notes, "and we have seen a number of duplicators switch over to the 'chrome chain' to make that possible at economical high speeds of up to 64:1. By using chrome running masters, they are finding that it's possible to transfer the original signal to chrome cassettes with greater fidelity that cassettes duplicated with ferric tapes at much slower and costlier speeds."

BASF's Loop Master 920 is a backcoated chrome mastering tape specifically designed for the high speeds of cassette duplication. With 920, BASF claims 7½ ips master quality can be achieved at 3¾ ips master speed, because the master uses higher coercivity chromium dioxide rather than ferric oxide. "With ferric loop bin tapes at 64:1 duplication speeds, the dynamic range is actually *less* than the chrome cassette tape onto which it's recorded," O'Kelly explains. "The master, in other words, is less dynamic than the cassette, and this puts a severe crimp in the quality of the finished product."

### SOLID STATE LOGIC REPORTS ORDERS FROM BBC, NBC, AND TOKYO GENZOSHO

• The British Broadcasting Corporation has awarded a contract for four SSL SL-5000M Series consoles to equip two of BBC Television's post-production suites. Sypher Suites 3 and 4, located in the

**NEWS** . . . *continued on page 157 —*

# RECORDING ORCHESTRAL, STRING AND HORN SECTIONS

## Achieving a Unique Blend of "Classical" Textures for Contemporary Sessions

*by John Kurlander*

The Royal Philharmonic Orchestra in Abbey Road Studio one, recording Laurence Leonard's *Mezoon*. Photography by Paul Spencer

**John Kurlander, one of Abbey Road Studios' leading engineers, is experienced in many diverse areas of sound recording, ranging from rock to classical. Many of his projects include orchestral work — often with a rhythm section — and recording orchestral forces outside the "conventional" classical field often requires an unconventional approach to the art. This article dicusses the techniques Kurlander has applied to four specific projects, several of which were nominated for Grammy Awards and reached Platinum sales levels: the Original Broadway Cast recording of *On Your Toes; Hooked On Classics (Volume 1); Toto IV;* and Andrew Lloyd-Webber's *Requiem.***

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In this article I describe how we built an individual, personalized sound style for each of four, very different projects. The room layouts and instrumentation line-ups are also included here for reference, but I have no intention of boring you with the minutiae of recording techniques, mike positioning and EQ settings. Much more to the point are the, perhaps, unconventional aspects of each project —the special, customized elements of each sound.

### *ON YOUR TOES* BROADWAY CAST RECORDING AT RCA STUDIOS, NEW YORK

In 1983, the Rogers and Hart musical *On Your Toes* was revived on Broadway — not as a modernized production, but aiming to stage as closely as possible a faithful reproduction of the 1936 original. The Thirties musical arrangements, by Hans Spialek, were used along with the original instrumentation and vocal line-up; not one deviation or addition

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would be tolerated by John Mauceri, co-producer/musical director. Consequently, the brief for the cast album recording demanded a "true" 1936 sound, captured digitally!

We were fully aware that the majority of Broadway cast albums normally are recorded using analog multitrack techniques, often handling some of the vocals as overdubs and taking weeks of studio time to complete. Doing it this way meant that the project had to be a particularly long-running, successful show; otherwise the majority of the major record companies couldn't justify the immense budgets required.

The session's executive producer John Yap, insisted that *On Your Toes* be recorded direct to two-track, using a JVC BP-90 digital stereo processor, with no overdubbing, and that the entire double album should be recorded in four consecutive sessions. As a result, we were able to record on Monday, edit the tapes on Tuesday,

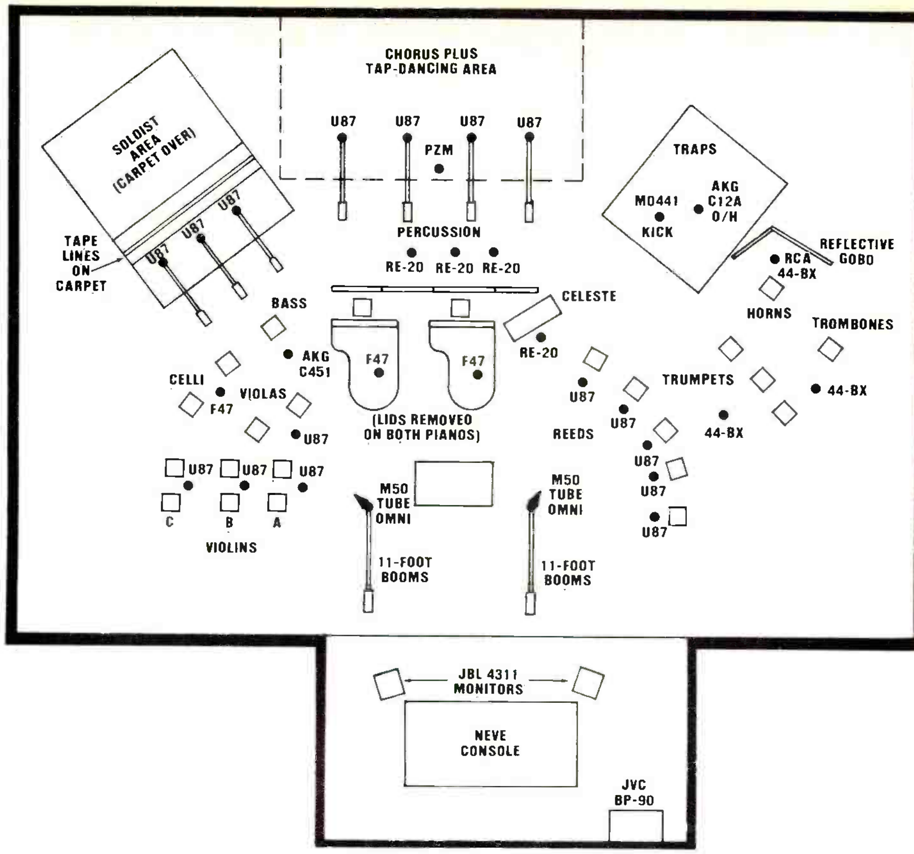


**Figure 1: Room layout for *On Your Toes* sessions at RCA Studios, NY**

John Price

— **the Author** —

John Kurlander joined London's Abbey Road Studios as a Junior Assistant in 1968 at the age of 17. After working in the tape library for a few months, he became a tape-operator and some of his first sessions included The Beatles' *Abbey Road* album. Later, he worked with John Lennon, Paul McCartney and Ringo Starr on solo album projects. While working with EMI house producer Norman Newell, Kurlander became an engineer, and the pair have since recorded numerous albums, ranging from Geoff Love to Broadway shows.

While he has also worked extensively in the classical field — recording Riccardo Muti, Seiji Ozawa, Eugene Ormandy, Klaus Tennstedt and others with many of the world's leading symphony orchestras, including the Philadelphia and Berlin Philharmonic — in the contemporary field Kurlander has also worked with Elton John, Toto, Olivia Newton-John, Jeff Wayne (on the highly successful *War of the World's* album), and many others. Most recently, the engineer has been working with the London Symphony Orchestra on a new *Classic Rock* project at Abbey Road, and with top classical producer James Mallinson in Paris on a project with Seiji Ozawa.

and play the completed recording to the client on Wednesday. Not only were the budget considerations more than satisfied, but John Mauceri was also totally in favor of keeping the whole cast together, and recording the show as one integral whole. With everyone, cast and technicians alike, working at performance pitch throughout the sessions, we achieved an element of theatrical spontaneity and togetherness not usually found in cast recordings — a particular *forte* of the producer, Norman Newell.

The recording went like a dream; the album subsequently received a Grammmy nomination, and was voted "Best Cast LP of the Year" by *Stereo Review*.

My own involvement started about two weeks before the sessions, listening to rough tapes of the show taken from the theater's PA System. Then Norman and I went to see the show several times, to absorb the atmosphere and soak in the score — when you are mixing straight to stereo, you *have* to know the score inside out. During this preparation period I planned the studio layout, drawing up plans and agreeing on them with both Norman Newell and John Mauceri. When you intend to work tour sessions in one day, it is vitally important to finalize everything and cover as many eventualities as possible, well before the orchestra starts to tune up!

Figure 1 shows the layout of RCA Studio C for the recording session. The instrumental line-up was as follow: two pianos with the lids removed (one pianist doubling on celeste); five reeds, all doubling sax, flutes and oboes; one french horn playing into a reflective screen; three trumpets, one trombone; one percussionist (playing tymps, xylophone, wood-block, bells, etc.); and "traps." (Traps are basically the old-style drumkit — kick drum, snare, hi-hat and two toms —

. . . continued on page 36 —

**Recording the sound of tap dancers during the *On Your Toes* session using a floor-mounted Crown PZM microphone.**

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# ORCHESTRAL RECORDING
## Blending Classical Techniques with Contemporary Styles

— continued from page 33 . . .

which were placed in RCA's movable drum booth.) In addition, there were six violins, grouped in pairs as A, B and C; two violas, two celli, and double bass. The line-up was completed by soloists, chorus and tap-dancers.

The layout was kept as symmetrical as possible. Apart from the drum booth and the reflective screen for the french horn, no acoustic screens or gobos were used.

The important aspect of the setup is a pair of Neumann M50 tube omnis placed approximately 10 feet apart and 11 feet high, on either side of the conductor's podium, accounting for 60 to 70% of the orchestral balance. The M50's gave me the room sound that is so characteristic of Thirties recordings. In addition, I fed in (at quite low level) a selection of close mikes, each chosen specifically for the sound they would contribute to the main omnis — for example, an AKG C451 on bass would not be chosen as a bass mike on its own, but since the full sound of the instrument projects up to the omnidirectional polar patent, a close mike is only needed to give that extra "edge" and close "bite" missing from the overheads. Using this technique, the stereo picture on the record is not unlike that shown in the layout diagram, with the exception that the vocals are panned into the center.

The solo vocalists were discouraged from using the mike too closely, to achieve a clearly theatrical performance rather than the more usual "studio-recorded" vocal sound. Capturing this true "theater sound" from the tap-dancing chorus promised to be more of a problem, since taps are notoriously difficult to record with enough dimension. Norman took this problem very seriously, and brought in a dancer some days before the session to experiment with different techniques. Our final solution was to use a Crown PZM — which proved particularly good for transients and high frequencies — mounted on the wooden floor in front of the dance area, panned center, to supplement the vocalists' Neumann U87s, which gave a really "dynamic" feel to the tap dancing routines.

In the control room, I arranged the soloist, chorus and tap dancing mikes to come up near the middle of the board, and sat in front of the stereo subgroups. In addition to the orchestra, rhythm and overheads, I stereo subgrouped the soloists, chorus/dance mikes, and echo returns. For show albums, especially where there is a need for a "period" feel, I stay away from digital reverberation, and use

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Pictured in RCA Studio C during the *On Your Toes* dates (*L-to-R*): Joaquin J. Lopes, Gil King, Paul Schwartz, John Yap and author John Kurlander.

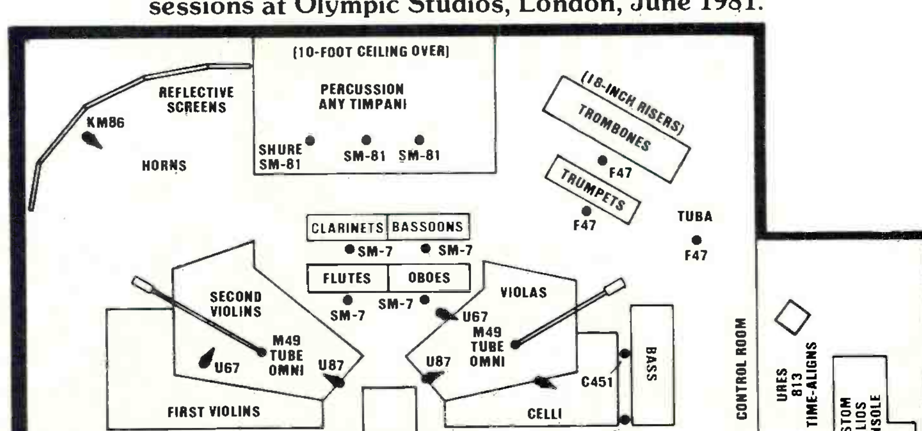
the stereo EMT 140 plates. In this case I used a tape delay in the send, set at around 100 to 150 milliseconds. I tend not to use a great deal of reverb, in an attempt to keep some similarity to the feeling you would get in the theater. I used no compression or limiting at all on the recording, although George Marino at Sterling Sound, New York, added a very small amount of subtle compression on the final vinyl cut. (The Compact Disc, however, has no overall compression.)

The whole setup was handled very much the way you set up for a live mix. The day before, I got the assistant to move around the room, talking into all the mikes, and got my basic levels and echo settings. When you have a four-session day — from 9:00 am to midnight — you can't leave any of the set up for the morning. If there is any one "secret" behind my recordings, it *has* to be preparation.

## *HOOKED ON CLASSICS* RECORDED AT OLYMPIC STUDIOS, LONDON; REMIXED AT ABBEY ROAD PENTHOUSE STUDIO, LONDON

*Hooked on Classics* satisfied one of my oldest ambitions: simultaneously to have a "classical" record at #1 in both the U.S. and U.K. Full credit for dreaming up the idea of Disco-classics lies with producers Don Reedman and Jeff Jarrat, following on the success of their *Classic Rock* series recorded in the late Seventies (with which I was also involved). . . continued overleaf —

**Figure 2: Room and microphone layout for *Hooked on Classics* sessions at Olympic Studios, London, June 1981.**



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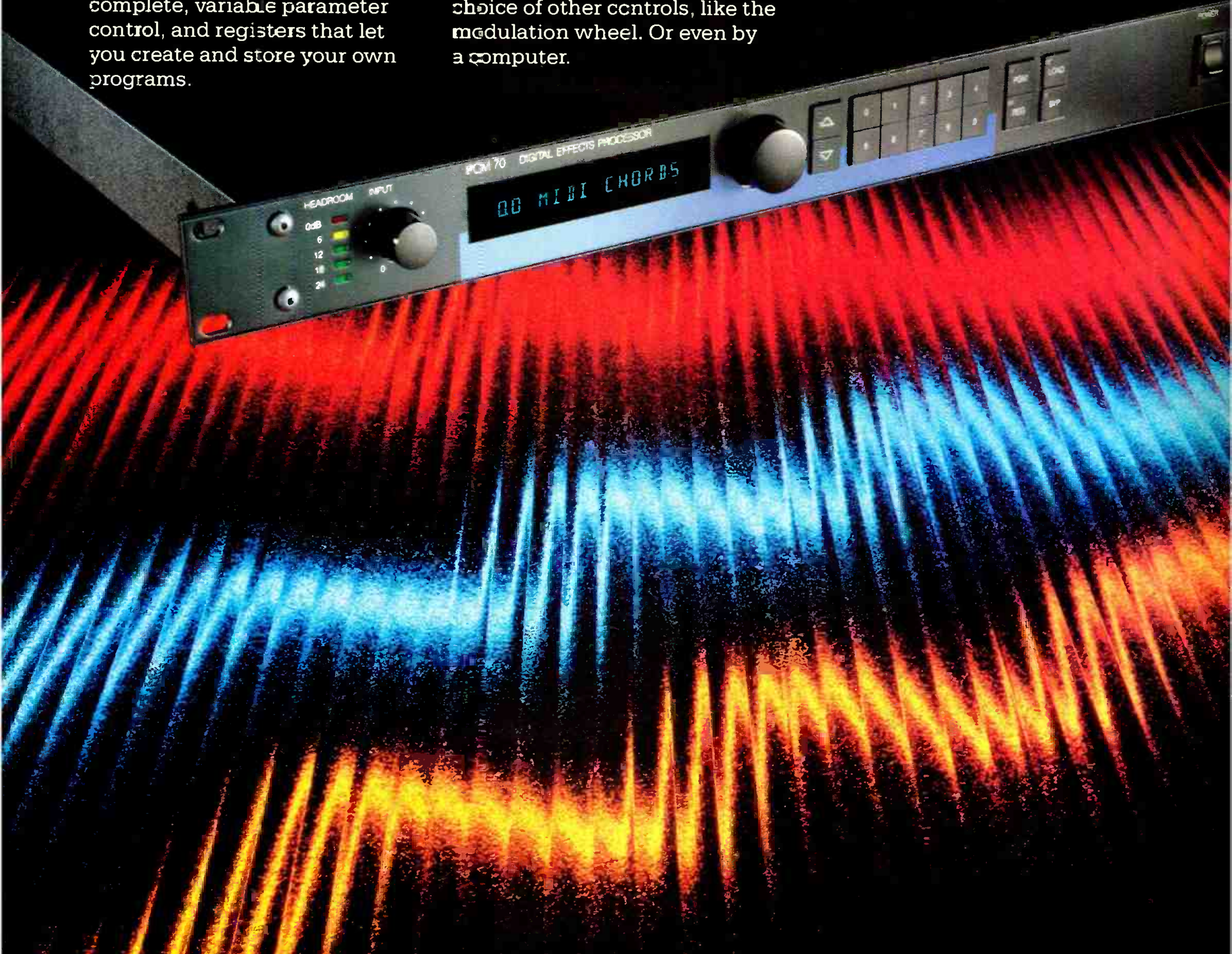


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Photo by Paul Spencer

Typical setup for Royal Philharmonic Orchestra string section


This project involved taking brilliantly devised arrangements by Louis Clarke, then a symphony orchestra —the Royal Philharmonic Orchestra —and, by adding the catalyst of a Linn Drum machine, producing a *totally* different musical entity. We wanted to create the illusion of a symphonic orchestral recording combined with a strong dance beat, something that is impossible to achieve by standard classical recording techniques. A symphony orchestra creates its own blend of tonal colors and dynamics while the drum machine simply ploughs a hole straight through the middle of these dynamics, shattering the whole blend. My approach was to take the classical dynamics to pieces, slot in the Linn, and rebuild the sound to create an illusion of simplicity.

Part of the secret also lies in the arrangements. You think you're listening to the original settings, because it sounds like them. But in fact there is a subliminal, subtle rhythm pervading each piece. When one part of the orchestra has the tune, another part will be playing a rhythmic figure that was never in the original piece.

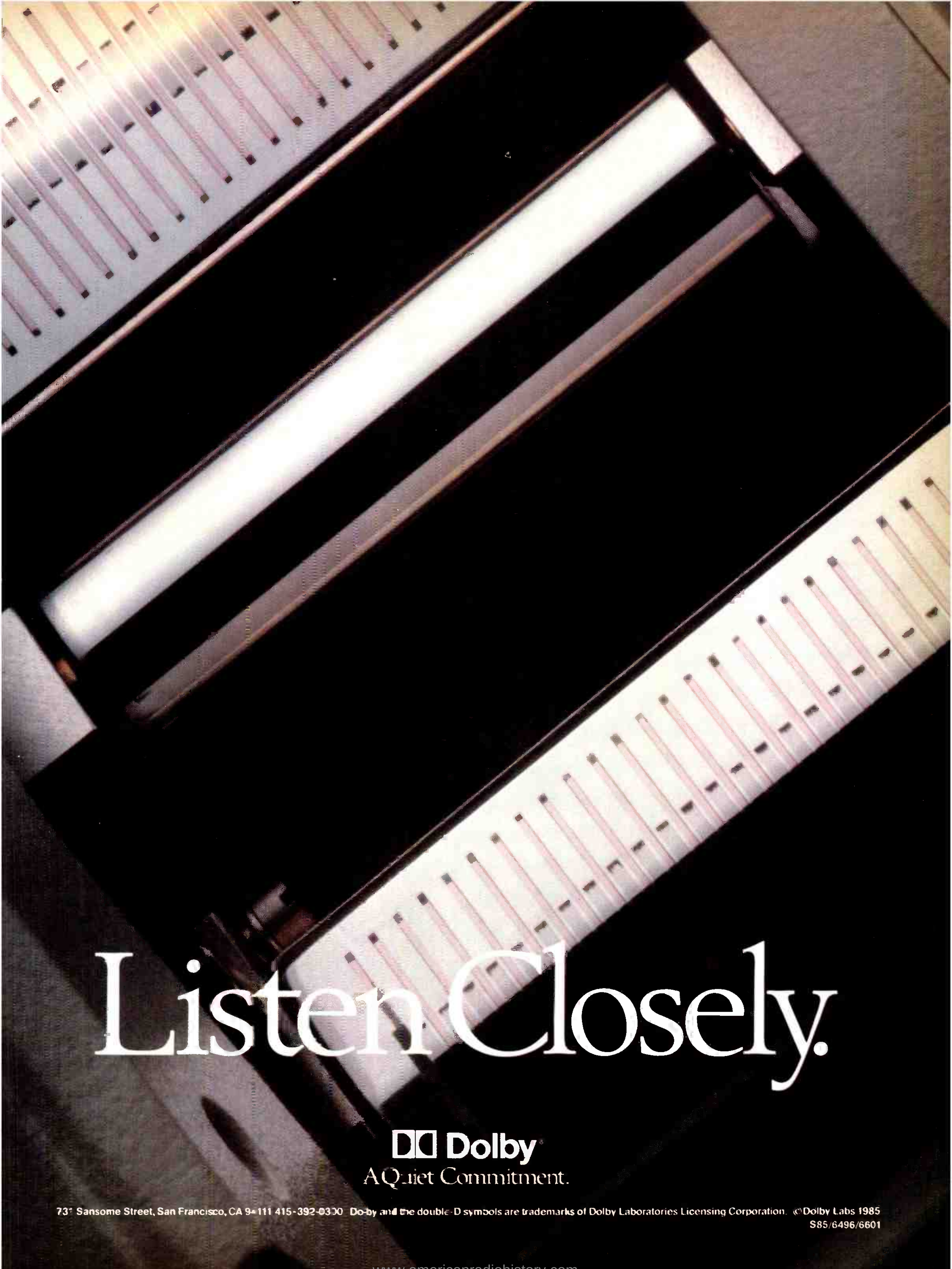
The line-up was bass guitar (played on some tracks by Herbie Flowers, and on others by Les Hurdle); Linn LM-1; 10 first violins; 10 second violins; eight violas; eight celli; four basses; two flutes; two oboes; two clarinets; two bassoons; five horns; three trumpets; three trombones; one tuba; tymps and assorted percussion; and various solo instruments. The basic room layout is shown in Figure 2. Immediately noticeable are the two sets of string mikes and the overheads, used to gain different perspectives on the orchestra. My approach lies in recording a number of different perspectives of the orchestra — in a similar fashion to recording drums with close mikes, overheads and perhaps some ambience mikes in the room. Each type of mike is selected to catch different aspects of the sound, therefore avoiding the usual multi-mike/phase syndrome.

The four "inner" Neumann U87s look out on the strings at a height of about nine to 10 feet. On the outer desks, pointing in different directions at a slightly lower height, are the U67 tube mikes, with a different sound and different EQ. The inner set were panned closely to center, with the outer quartet panned across from hard left to hard right.

The overheads M49 tube mikes set to omni were brought in on large boom stands from the back at about 13 to 14 feet, so that they would cover the whole orchestra. Which completes the three perspectives referred to above; in the mix, each set of perspectives is equalized, compressed or expanded as appropriate, to give a sound that will cut through.

The mikes were set up as if the whole orchestra was to be recorded together. We started off with the Linn and bass guitar, recorded DI on five tracks, plus Linn sync code on track #24. Then, dividing the orchestra into "loud" and "soft" instruments, we first overdubbed the strings and woodwinds together, and then (leaving the string mike open) we recorded the brass and percussion. The final overdubs were solo instruments, recorded individually: harp, piano, guitar, solo violin, and so on.

During the mix, all the string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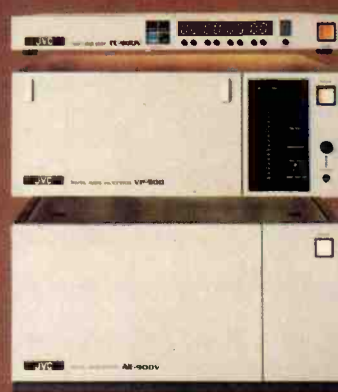

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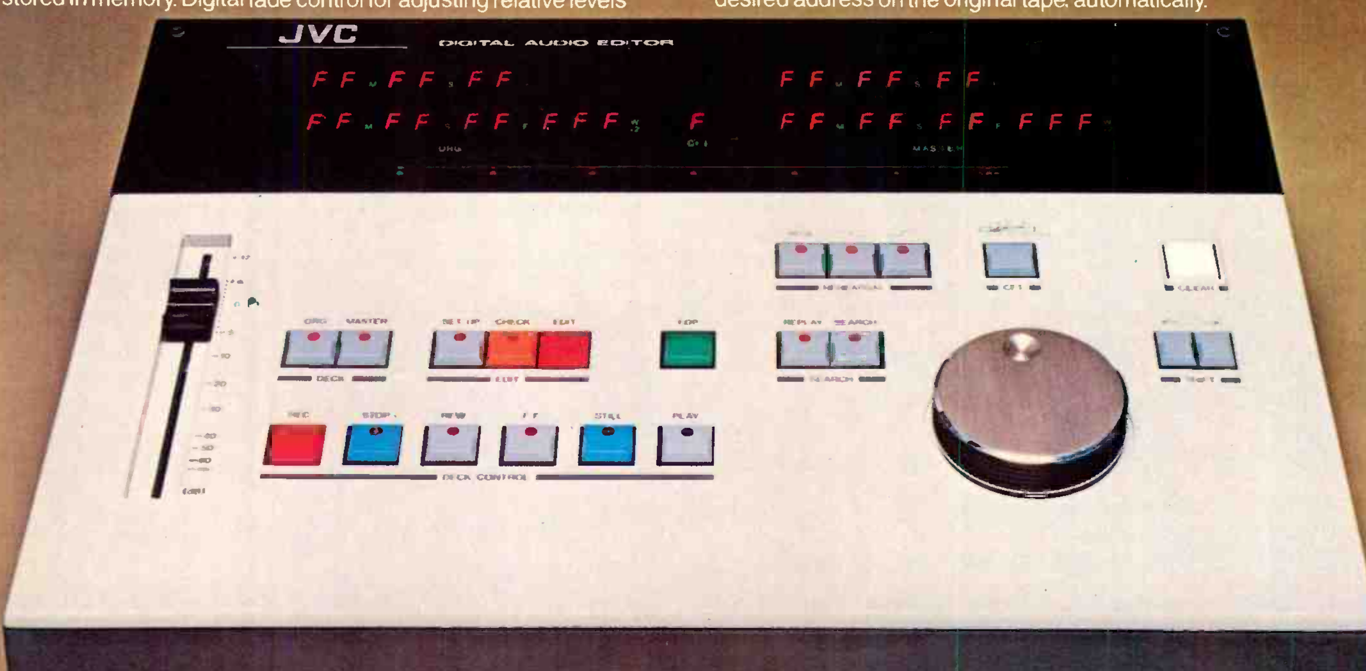


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tracks, both the overheads and the close mikes, were limited. While the individual tracks had largely been equalized during the recording, the mix as a whole — with the exception of the echo returns — was equalized to make it sound more like a dance mix. As each track progressed, we used different amounts of the echo returns, which were unaffected by the overall EQ. The Linn, with its very basic "boom-cack" rhythm, was placed quite high up in the balance, again to emphasize the dance beat. We then did A/B comparisons with various club mixes, to achieve the right amount of "edge" to the overall sound of the mix.

Because, above all, *Hooked on Classics* was a dance record, the most important thing was to get people's feet tapping, and to project the atmosphere of infectious good humor out of the sessions and onto the disk.

Peter Vernon

**Pictured during the *Hooked on Classics* sessions with the RPO (*L-to-R*): producers Jeff Jarratt and Don Reedman, with engineer John Kurlander.**

## *TOTO IV* ORCHESTRAL OVERDUBS AT ABBEY ROAD STUDIO ONE, LONDON

When James Newton-Howard was commissioned to arrange and conduct the orchestral overdubs for *Toto IV*, he recommended the use of Studio 1 at Abbey Road as being clearly the best studio in the world for orchestral recording. I had worked with James on similar orchestral overdub projects for Elton John (*Blue Moves* and *The Fox)*, and we both understood the problems involved in matching our Abbey Road-sound to the original tracks.

Since the orchestral overdubs were only a small contribution to the total project, an understanding of the album's total sound was deemed important to enable the orchestra tracks to slide smoothly into the complete mix. However, we were working with slave multitrack tapes, and were very vague about the shape that the final mix would take.

I worked around this problem by providing David Paich with three separate stereo tracks to play with, each track being a complete entity in itself, but also compatible with the others to allow intermixing to produce exactly the desired final effect.

The line-up on the date was 16 first violins; 14 second violins; 12 violas; 10 celli; and 8 basses — refer to Figure 3.

The first stereo pair comprised an AKG C24 placed very high (approximately 18 to 19 feet) over the conductor's head, optimizing the 2½-second natural reverb time in Studio 1. The second stereo pair was two Neumann M50 tube omnis mounted directly over the musicians, enhancing the upper harmonics of the strings, while the third stereo track was derived from a group of five Neumann U47 tube mikes from Abbey Road's enviable vintage-microphone collection, which gave an unmistakably "ballsy" sound. All the microphones chosen for this project were tube models; any disadvantages resulting from poor signal-to-noise ratio was far outweighed by their total suitability for this particular application.

*. . . continued overleaf —*

NELS

**At the *Toto IV* string overdub session (*L-to-R*): Marty and David Paich, engineer Kurlander, and arranger/conductor James Newton Howard.**

## *REQUIEM* FULL ORCHESTRAL AND CHOIR RECORDING AT ABBEY ROAD STUDIO ONE; REMIXED AT CAPITOL STUDIO C, L.A.

The experience gained while working on *Hooked on Classics, Classic Rock* and Broadway shows was a positive benefit when I was asked to become involved with Andrew Lloyd-Webber on the debut recording of his *Requiem,* in association with engineer Martin Levan who, as well as handling all of Andrew's theater PA sound, has also recorded some of his more recent work. For this project, Andrew wanted a good classical recording to satisfy a discerning classical audience, while at the same time appealing to the wider "pop" record-buying public.

Since *Requiem* had never been recorded before, the line-up, arrangements and even the score changed from day to day during the three-week period between the first rehearsal and the final recording sessions. I was involved from the first rehearsals, which gave me the opportunity to understand the project more fully, and to do the necessary preparation work that always pays off "on the day."

As the line-up changed, so did the planned studio layout. The final room layout, Figure 4, shows the choir positioned behind the conductor, and the soloists directly in front of him. At the first rehearsal, the choir was placed *behind* the percussion, but the conductor Lorin Maazel felt that since all the voices formed such an integral part of the whole work, he wanted them closer for greater communication. We discussed this together at length, agreeing on the final layout.

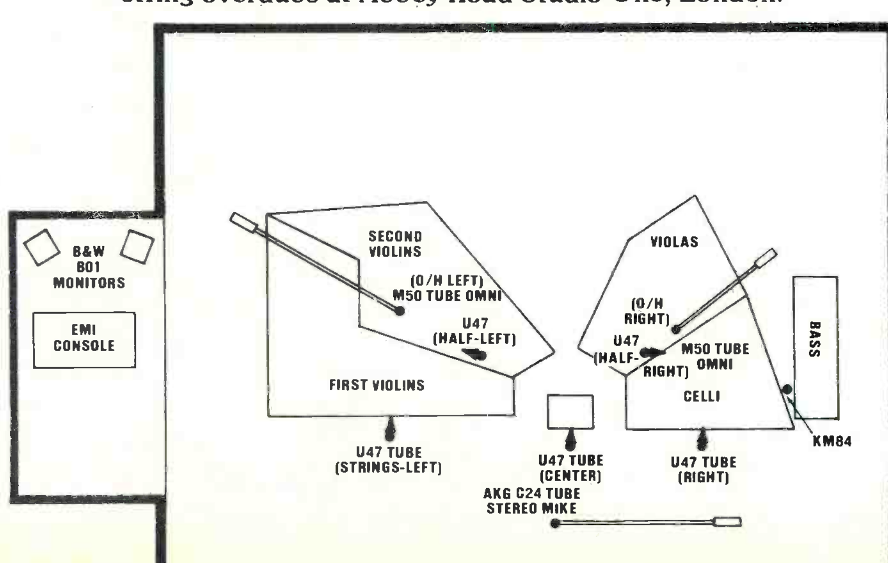
The instrumentation line-up was as follows: 12 violas; eight celli; four basses; two flutes (one doubling piccolo); two oboes (one doubling Cor Anglais); one contra-bassoon; three trumpets, three trombones plus one bass; four french horns; one baritone sax (doubling on alto); two clarinets (one doubling on E-flat, and the other on bass clarinet); plus harp, piano, celeste, tympani, percussion, synthesizer and an Allen Computer Organ. There were also three soloists: Placido Domingo (tenor); Sarah Brightman (soprano) and Paul Miles-Kingston (treble), with the Winchester Cathedral Choir directed by Martin Neary.

One instrument, in particular, posed an interesting recording problem. The Allen Computer Organ is designed to simulate the sound of a church organ, and has about a dozen spacers, split into eight for the mid/high registers, and four for the bass. I miked the mid/high speakers conventionally, directing them into a Neumann U87 and panning left to achieve the enormous impact of a church pipe organ. To record the bass registers, we decided to bring Studio 1's own acoustics into play by placing the low-frequency speakers on the floor facing into the corner, and allowing the enormous low-end "boom" to be picked up acoustically by ambient mikes only.

In addition to the 30+ instrument and vocal mikes, a submixer was used to handle the DI inputs from synthesizers and a Simmons Electronic Drum Kit, and an umbrella placed over the kit to gently reduce sound spillage.

I'd successfully used the Calrec Soundfield microphone as a stereo mike on some previous classical recordings. I brought it in as the main stereo pair, set to figure-eight mode, at a height of about 16 feet over the conductor's head. Because I wanted to get a very open sound on the voices, the singers were placed in a prime position to enable the inner viola and celli mikes to pick up some singing, along with the Soundfield, and open out the sound. There were a couple of high overheads — Schoeps omnis with MK5 capsules — placed halfway be-

**Figure 3: Room and microphone layout for *Toto IV* string overdubs at Abbey Road Studio One, London.**



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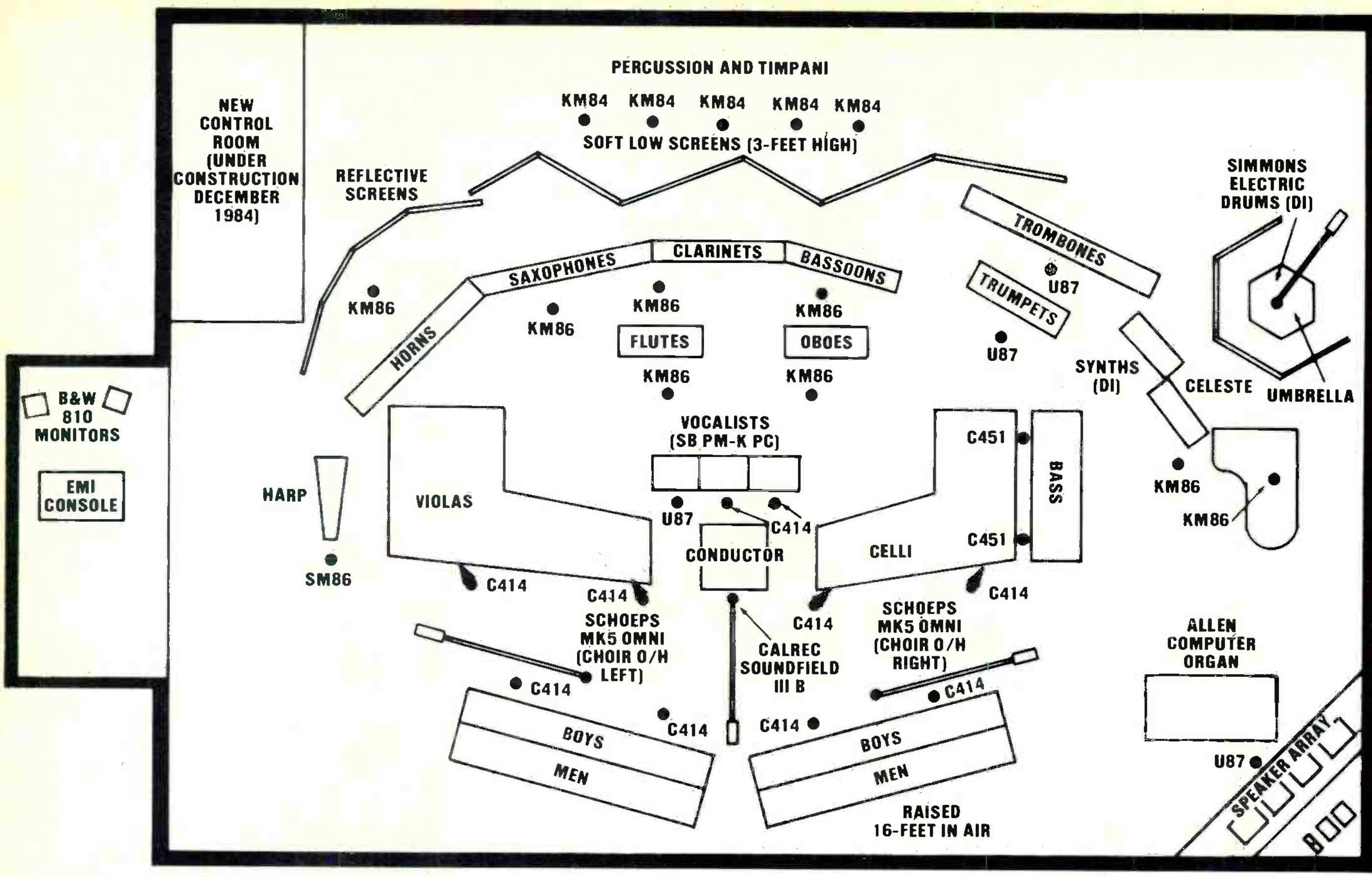



**Figure 4: Room and microphone layout for *Requiem* orchestral sessions at Abbey Road Studio One.**



tween the choir and the orchestra, so that they picked up ambience from the strings and voices (mainly the men). On the soloists, we found that a U87 suited Sarah Brightman, while the AKG C414 worked well for Placido and Paul.

Again, the orchestral spot mikes here were largely designed to fill in areas of the sound that were not picked up effectively enough by the overheads — you don't need a "complete" sound on every one. In some respects, this was even true of the voices. Placido commented that he liked the sound we captured on his voice, and this was largely because it was being picked up by the Soundfield mike as well as the C414.

The track layout was basically stereo strings, stereo woodwinds, stereo horns and brass, stereo percussion, stereo close choir, stereo overhead choir, and then organ, keyboards, harp, Simmons, and so on. I tend to put down stereo tracks; for example, if I was tracking stereo woodwinds, I would pan them across between extreme left and right, and reduce the width as necessary during the mix.

We had originally planned to remix *Requiem* at the Penthouse Studio, Abbey Road, where I've tended to mix most of my "crossover" recordings over the past few years. But, for various reasons — including the fact that


Clive Barda/EMI

*Requiem* session with English Chamber Orchestra and Winchester Choir.

Andrew needed to be in Los Angeles for the opening of *Cats,* and to avoid mixing over Christmas — we used Capitol Studio C, L.A. The facility's Neve Model 8108 gave us a sound not unlike what we would have achieved with the Neve in the Penthouse Studio at Abbey Road. I like the natural, open sound of the Neve 81-Series, especially at the top-end, and it seems to work well with material recorded in Number One Studio.

We experimented with some digital reverbs, and Martin knew the kind of vocal sound and echoes that Andrew liked. We ended up using the AMS RX-16 and Yamaha REV-1 reverb systems, adding reverb to the orchestra and to the vocals and choir — particularly the latter. We tried to help the overall effect into a church-like acoustic, although we didn't want to try and fool people into thinking it really was done in a church.

In terms of an overall balance, the strings were considered very much a bedding, while the soloists, choir and organ were the crux of the balance, with a lot of brass and heavy percussion going on. Andrew was very much involved in the mixing process. We spent a day at Capitol before he arrived, acquainting ourselves with the control room and monitoring, doing a rough mix and playing it back in another room, and so on. The next day we did a rough mix of about 10 minutes of the material, and played it to Andrew for comments. He felt there was too much reverberation, and wanted more clarity on the voices, so we altered the reverb texture and level. He instantly loved it, and after that everything fell into place.

David Murray, the producer for EMI, chose the take to use for each section, and we mixed that along with any edit inserts. The whole of Side One was mixed in five hours, and we then spent the evening editing. Andrew approved the edits the next morning, and in the next six hours we mixed the rest. Once we'd got the balance, it was just a mater of zipping through it.

We very much felt that we should try to mix the session continuously, rather than concentrate on small sections bit by bit — an approach that paid off. After the final editing and playback, there was just one small part and a couple of edits we wanted to re-do — in essence, the entire mix took just 12 hours.

Both EMI and Andrew wanted to have plenty of dynamic range — EMI specifically asked for "an incredible CD" — and by using the Sony PCM-3324 digital 24-track and mixing to Sony PCM-1610, we had plenty of headroom.

The secret to *Requiem*, I feel, was very much in the preparation. By and large, once we got into the studio, the session was very easy. Preparation, and going with the mood of the people who are making the music, is very important, whatever the project. No doubt there are certain common elements in the way I work, but I place a good deal of importance in matching the technique to the music, and the facilities available. ■■■

# DESIGN EVOLUTION OF TWO-BOX PACKAGED LOUDSPEAKER SYSTEMS

## Sundown Sound and Audio Techniques Follow Similar Design and Development Paths in Pursuit of an "Ideal" Speaker Package

All photography by David Scheirman

*by David Scheirman*

In the ever-changing concert sound business, one of the most often-discussed topics is the packaging of loudspeaker systems. The method decided upon by a particular sound company to house, transport and deploy the system's loudspeaker devices will determine the amount of labor and truck space required, the longevity of the system in terms of both popularity and profitability, and will dictate the options available to technicians when assembling large arrays with the particular enclosures. Another important consideration is the actual sonic characteristics of the system when modular boxes are combined in groups of like kind.

Within the past year, two different sound reinforcement firms with no relation to one another, working independently and with the help of nationally-recognized audio consultants, have developed two sound system packages for touring use that are strikingly similar in some respects. The fact that these two loudspeaker systems have appeared on the touring scene at the same time for use by smaller, aggressive sound rental firms is perhaps indicative of an interesting trend. A close look at these two companies and the methods each used to arrive at the design of a new sound system follows.

### Two-box Touring Systems: A Brief History

In the early Seventies, pioneering tour sound firms such as Clair Bros. (Lititz, PA), McCune (San Francisco, CA) and Northwest Sound (Portland, OR) each developed touring sound loudspeaker packages that consolidated into one cabinet all of the various low-, mid- and high-frequency transducers required for a portable, full-bandwidth sound reinforcement system. These "one-box" systems, such as the Clair S-4 and McCune JM-3, helped to establish the concept of a *modular* touring sound system, with each cabinet being identically-sized and housing the same components.

By the late Seventies, contemporary music had begun to incorporate much more low-frequency information. The proliferation of keyboard synthesizers and electronic drums began to push some touring sound companies toward systems capable of offering extremely high sound pressure levels in the lower frequencies. Well-known touring firms such as Maryland Sound (Baltimore, MD), Showco, Inc. (Dallas, TX) and Stanal Sound, Ltd. (Kearney, NE) each introduced their own version of a new "two-box" system that had extended low-frequency capabilities due to the greater number of bass transducers offered with each compression driver when compared to the older "one-box" systems. Those companies using one-box concept developed accompanying subwoofers but, in many situations, the two-box systems were generally better able to supply a fuller bass sound with fewer cabinets.

Of course, such concurrent developments as increasing power amplifier headroom, and improved power-handling capacity of new loudspeaker units played an important part in the direction that the evolution of these various systems took. Today, nearly all of the aforementioned system configurations are still in use, although a broad range of improvements has taken place in many areas, including transducer technology and enclosure fabrication materials and techniques. Sound companies that are new on the scene continue to experiment with both one-box and two-box systems.

The success of modular loudspeaker packages in providing consistent, high-quality sound for live performances with a minimum of logistical effort inspired several different manufacturers to introduce their own line of enclosures for purchase by any firm or group requiring a loudspeaker system for performance use. Within the past five years, companies such as Meyer Sound Labora-

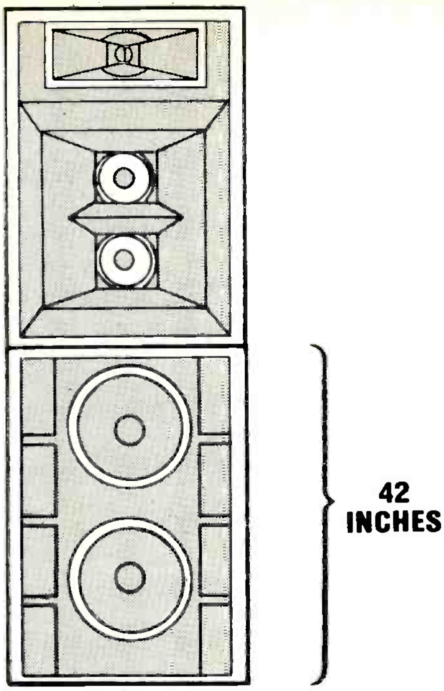


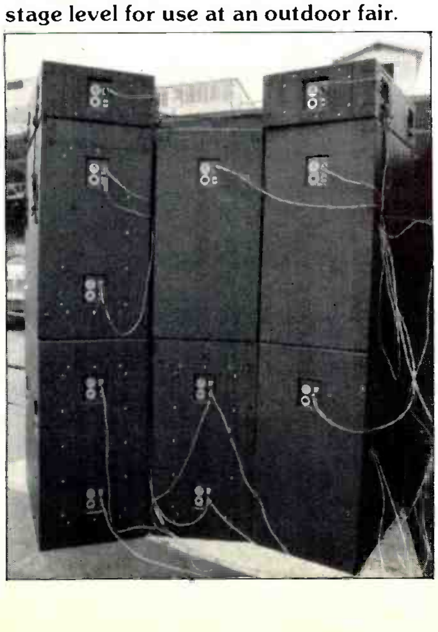

Figure 1: Sundown Sound's CDI "two-box" system.



tories, Turbosound, and Eastern Works have had a certain measure of success with factory-manufactured enclosures; most have been of the one-box variety, with subwoofers also being available. Other firms such as Community Light & Sound, Harbinger and Renkus-Heinz have also joined the race.

One manufacturer that has been quick to sense the market need for professional two-box speaker systems is JBL. In 1985, JBL Professional Products Division introduced its Concert Series™ line, which offers completely-packaged crossover/amplifier/speaker systems for rental and touring use. Design input from Stanal Sound, Ltd. and Electrotec Productions was instrumental in the development of the new line.

Figure 2: The SDS 1000 and 2000 enclosures are trapezoidal in shape, allowing the easy assembly of tight, compact arrays. This view from the back shows the identically-sized cabinets stacked at stage level for use at an outdoor fair.



It is readily apparent that the pioneering efforts of concert-sound firms help bring about the establishment of new product lines for established manufacturers. Since the new trend toward two-box systems has been spearheaded by such companies, let's take a look at Sundown Sound and Audio Techniques, Inc., each of which has recently developed a new two-box system that features a direct-radiating bass box housing a pair of 18-inch speakers, and a mid/high box that relies on mid-bass, horn-loaded 12-inch speakers. Each company used the outside services of an audio consultant for assistance in the design process.

## SUNDOWN SOUND

Located in Portland, OR, Sundown Sound is an aggressive rental firm that has been making a name for itself with regional fairs for nearly 10 years; clients such as the Central Washington Fair, Jackson County Fair and southwest Washington Fair rely on the company for all audio needs. Sundown has also supplied full sound systems to the Oregon State Fair since 1975.

"The success of this company has been tied to Rock and Roll," explains Sundown Sound owner and president David Cutter. "We do over 200 one-nighter and regional shows a year. However, I decided to get into the fair market because the sound quality seemed to be so poor, and I wanted to help upgrade it. For some reason, many fairs can justify paying for the talent, but *not* for the sound. For that reason, audiences in the past have suffered; that is changing now."

In 1984, Cutter felt that the time had come to develop a new sound system designed from the ground up. "I started out playing guitar in a Rock band in the Sixties with a couple of University paging horns for a sound system," he recalls. "Ever since then, I have noticed that people *always* want to know why they can't understand the words of the singer!

"With today's technology, I figured that we should be able to put together something that provided very articulate vocals, and still was a full-bandwidth music system. The recent developments in measurement techniques, and the understanding that we now have of how important are such things as phase and power-band response, have given anyone who is interested the ability to build sound systems that were only a dream a few years ago."

Cutter enlisted the help of fellow

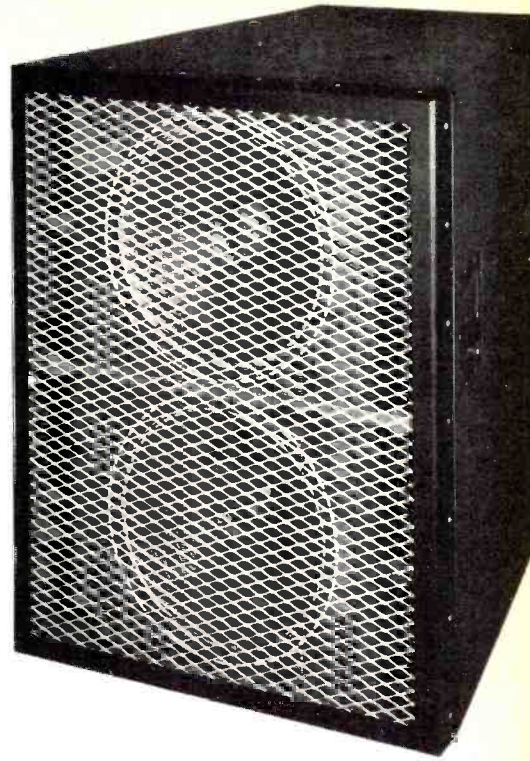
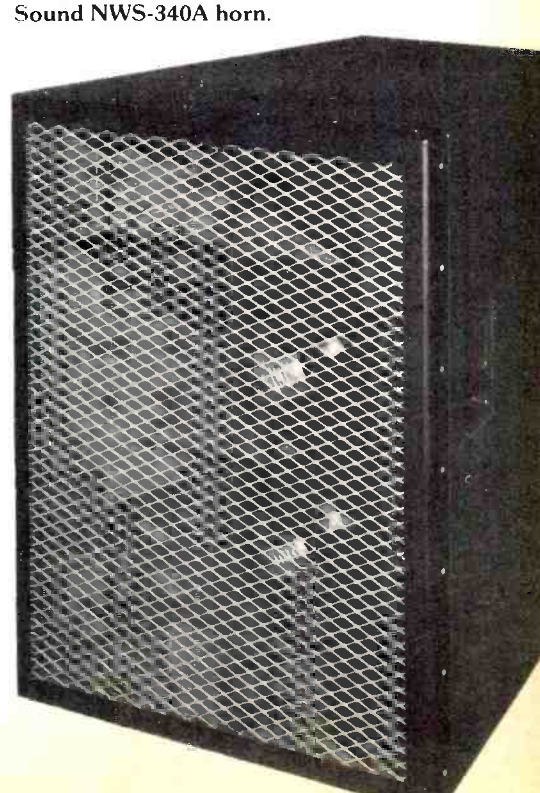


Figure 3: The SDS 1000 low-frequency enclosure houses a pair of JBL 2245H loudspeakers. The enclosure measures 42 by 34 inches, while the front face dimension is 34 inches.

Figure 4: The SDS 2000 mid/high frequency enclosure packs a pair of TAD TM1201 speakers and a TAD TD-4001 compression driver on a Northwest Sound NWS-340A horn.



Oregon resident Cal Perkins, and electrical engineer and audiophile whose contributions to sound system technology are many and legendary. (The old JBL Model 4560 vented bass cabinet is often known to users as a "Perkins box," for example.)

"David sat me down in his living room and played some Compact Discs over his hi-fi," recalls Perkins. "He told me that he wanted a portable sound system for rental use that sounded *that* good, from any seat in the house! It was a tall order, but we decided to give it a go."

Cutter and Perkins spent the summer of 1984 evaluating and selecting loudspeaker components. The result of their quest was Sundown's new CDI (Controlled Directivity Interface) two-box system (Figure 1).

"We chose that name to call attention to the fact that both the low and mid/high enclosures have complimentary horizontal and vertical polar patterns," Cutter explains. "You get consistent sound quality both on and off axis because the 'Q' of both types of enclosures is the same."

The cabinets are trapezoidal inshape, designed to form curved arrays with no gaps between the cabinets (Figure 2). "The pie-shaped spaces formed by rectangular enclosures form very effective mid-bass traps," Cutter notes. "Trapezoidal cabinets nest tightly against each other, doing away with that problem. The result is a cleaner, tighter, more coherent loudspeaker array."

### Low-Frequency Enclosure

The Model SDS 1000 low-frequency enclosure houses a pair of JBL 2245H 18-inch transducers that are loaded into a trapezoidal cabinet measuring 42 inches high by 34 inches deep; the front face measures 34 inches wide, and the back face 20.25 inches (Figure 3). This direct-radiator bass-reflex enclosure has a flat extended response below 40 Hz that is said to make subwoofers unnecessary. "The 3 dB-down point of the box is 38 Hz," Cutter states. "When you combine eight enclosures, that point moves down to 28 Hz. We have measured over 126 dB SPL for one cabinet at 35 Hz."

Each enclosure is fitted with a heavyduty front metal grill and a protective front edge frame. The cabinet's large vents run vertically down each side of the enclosure.

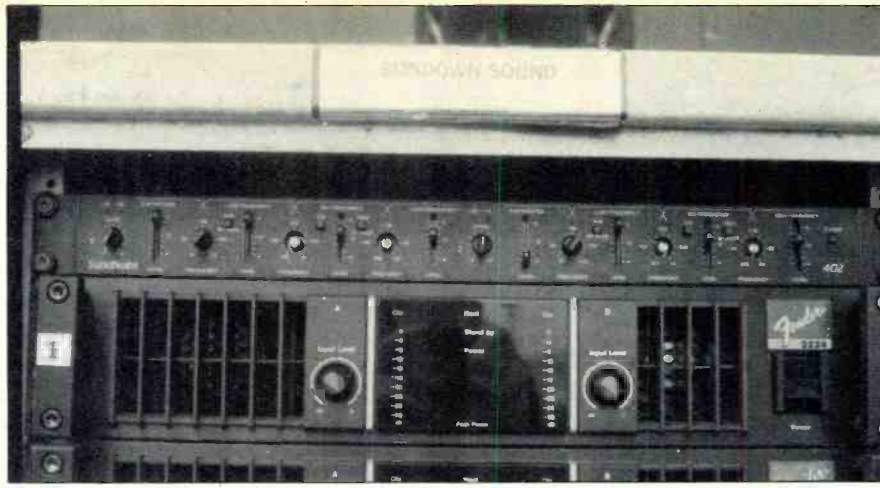


**Figure 5: A compact electronic crossover from Sundholm Electronics is included with each power amplifier rack. Crossover points are 200 Hz and 1.6 kHz.**

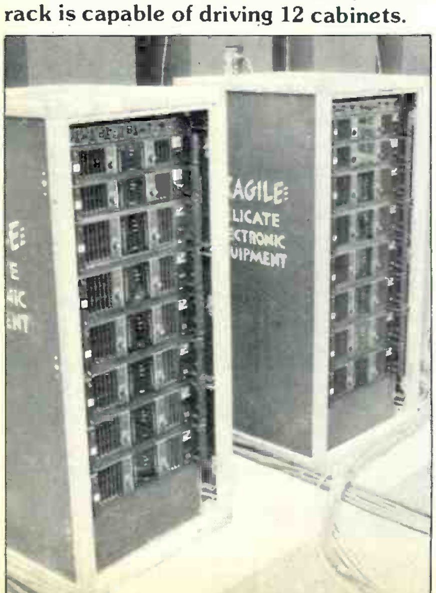


**Figure 6: Fender Model 2244 and 2224 power amplifiers are loaded into sturdy aluminum-frame racks for travel. Each rack is capable of driving 12 cabinets.**

### Mid/High Enclosure

The Model SDS 2000 mid/high enclosure is identical in size to the SDS 1000 (Figure 4). A horn-loaded midrange section houses a pair of TAD TM1201 12-inch loudspeakers with polymer-graphite cones. The enclosure provides a 90- by 40-degree, constant-directivity coverage pattern for the 12-inch speakers that is reportedly identical to the coverage pattern of the Northwest NWS 340A high-frequency horn. A TAD TD-4001 compression driver (two-inch throat four-inch diameter beryllium driver) completes the package.

The acoustic centers of both the mid- and high-frequency drivers are physically aligned so that no time delay occurs between the drivers in the critical crossover region. (A broadband 810-microsecond delay in the Sundholm electronic crossover aligns the acoustic centers of the SDS 1000 bass enclosure drivers within this same phasing plane.)

### System Electronics

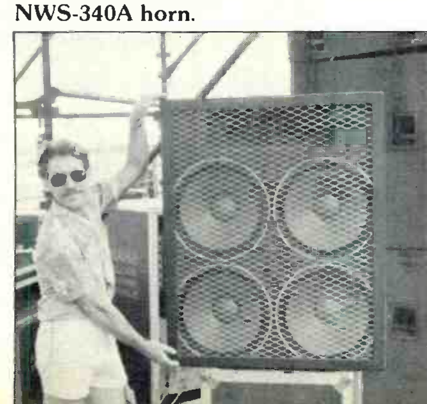
The Sundholm crossover, modified by Cal Perkins, is housed in each amplifier rack, usually located on stage behind the speaker stacks (Figure 5). "There has been much discussion about this point," recalls David Cutter. "When the actual crossover unit is located backstage in each amp rack, we have several spare units on all the time, ready to be jumpered over if there is ever a fault with one of the units. Also, it makes an ideal rental package when we are subcontracted to supply speaker stacks. We offer an engineered, neatly-packaged system, not just a pile of parts. No external electronics rack is required."

Amplifier racks fabricated for the system are sturdy, aluminum-frame assemblies that house a total of eight stereo power amplifiers (two Fender Professional Series Model 2224s, and six Model 2244s), in addition to the crossover (Figure 6). Each rack will power up to 12 enclosures: six bass and six mid/high.

### Additional Hardware

Sundown Sound also offers a full-range enclosure intended for use as a side-fill monitor or auxiliary fill cab-

**Figure 7: The SDS-400 SF is intended for use as a side-fill monitor or auxiliary main-fill speaker. The full-range cabinet houses four TAD TL-1601A speakers and a TD-4001 compression driver on an NWS-340A horn.**



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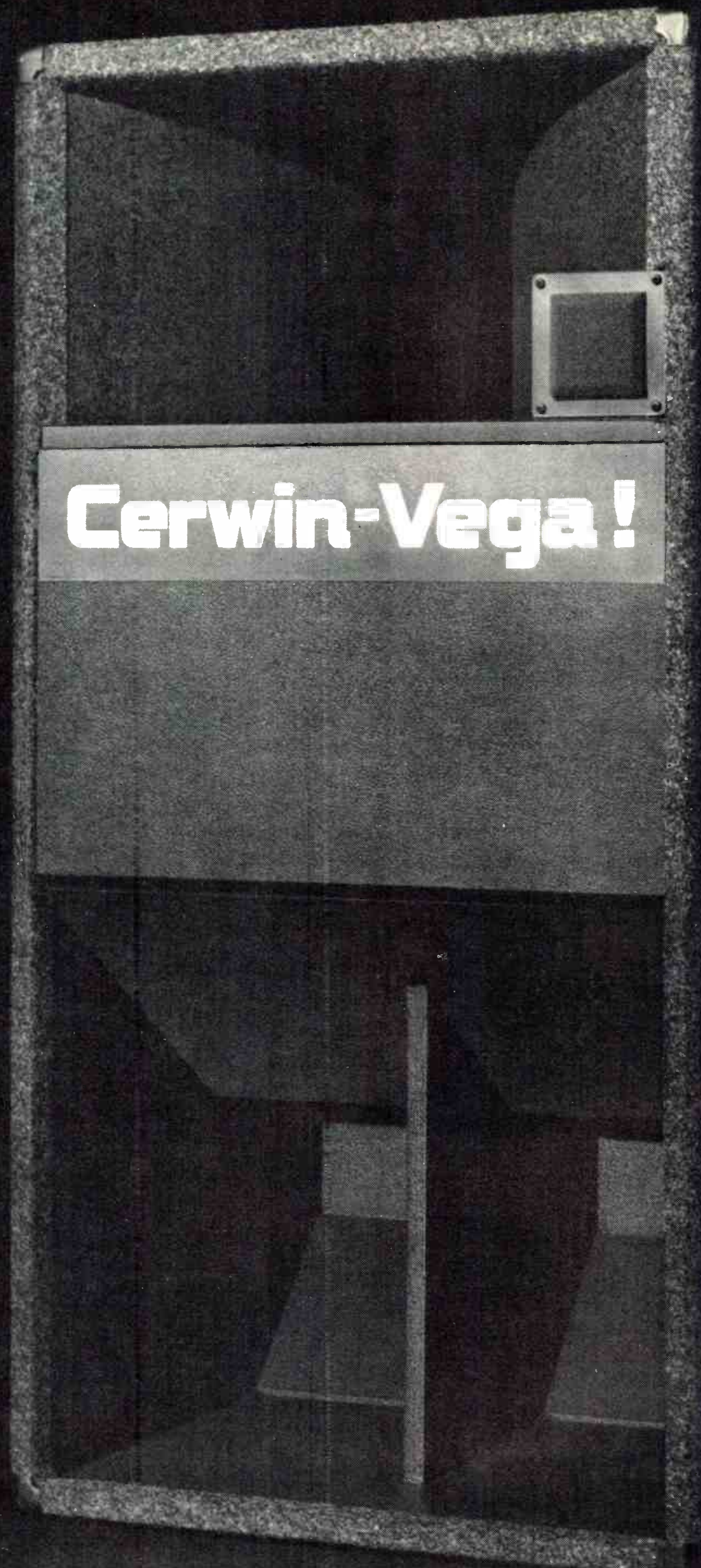


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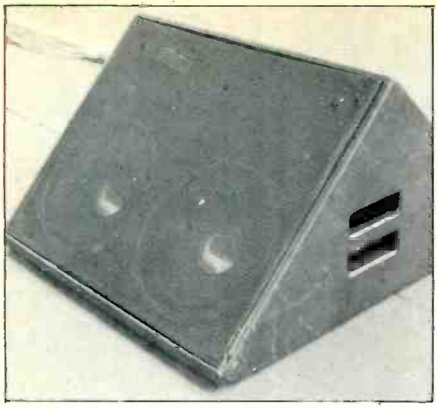

Figure 8: The SDS 200FM is a sealed-box, two-way speaker system for use as a stage monitor. Components are a pair of JBL 2202H speakers and a 2445J compression driver.



inet, the SDS 400SF (Figure 7). Identical in size and shape to the SDS 1000 and SDS 2000 cabinets, the 400SF houses four TAD TL-1601A 15-inch speakers along with a TD-4001 compression driver. The direct-radiating cabinet is said to be capable of producing over 130 dB SPL at 40 Hz, with a power-handling capacity (continuous program material) of 2,000 watts.

The company's standard floor monitor is the SDS 200FM (Figure 8). This sealed-box, two-way speaker system is loaded with two JBL Model 2202H 12-inch speakers and a Model 2445J compression driver loaded onto a NWS-340A flat-front radial horn. As with the house speaker system, 24 dB per octave Linkwitz-Riley crossover networks (packaged by Sundholm Electronics) are used, with custom time-compensation circuitry. Klark-Teknik third-octave graphic equalizers are typically supplied for each monitor mix (Figure 9).

Consoles are available from several manufacturers, including Midas and Soundcraft. "The type of console supplied is important to the people who actually have to use them," explains Sundown Sound technician Michael Johnson (Figure 10). "Many times we work with groups who carry their own sound personnel, but we do a lot of the mixing ourselves, too."

I had the opportunity to mix a show on the new Sundown Sound CDI system during the 1985 touring season, and was pleasantly surprised. The speaker system has apparently been designed so that typical "problem frequencies" do not require excessive equalization for the system to have an audibly "flat" response. Documentation supplied by Cal Perkins and Sundown Sound seems to bear this

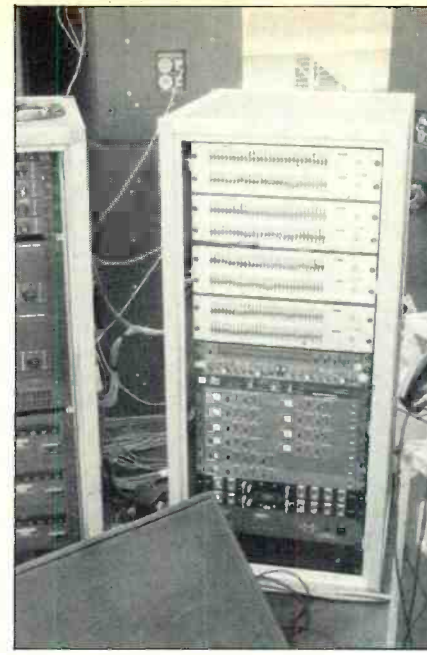

Figure 9: The monitor mix station includes Klark-Teknik third-octave graphic equalizers, one for each of eight mixes.

out, as shown by the frequency-response plot of Figure 11.

"The first time we turned this system on, I sat for several hours in disbelief," David Cutter recalls. "I had never actually heard music reproduced before from a speaker system that was so smooth and clean! At first I thought there was something wrong, but I realized that we had, after all, accomplished what we had set out to do: assemble a new system using the latest developments in audio "The Fender power amps can run full-tilt into a two-ohm load on a day when the

Figure 10: Sundown Sound technician Michael Johnson at the house mix position. A wide variety of gear is available, including the Midas console, dbx 900 rack and Klark-Teknik DN780 digital reverb.

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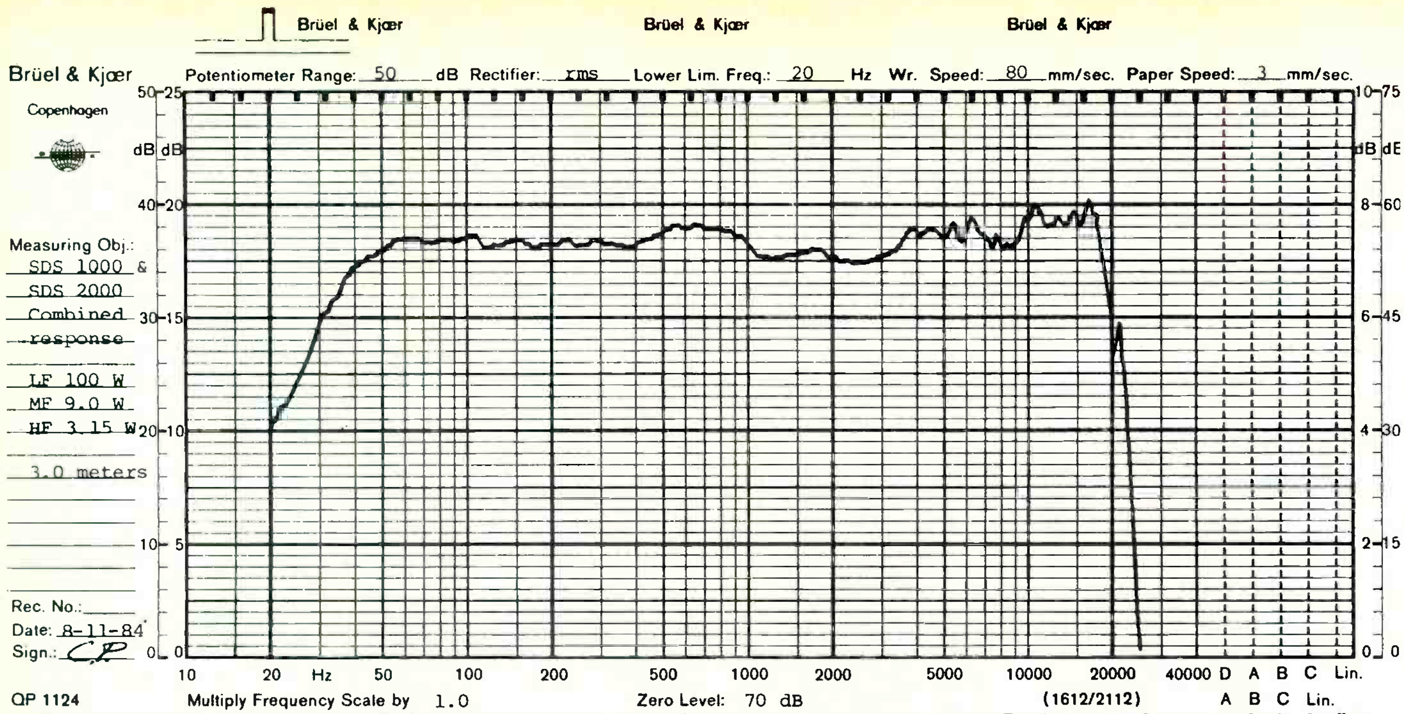


**Figure 11: Documentation supplied to Sundown Sound by electro-acoustical consultant Cal Perkins shows a relatively flat response for the SDS-1000/2000 combination, unequalized.**

temperature is 100° F, and still keep working. The TAD drivers have such a clean top-end that extra tweeters just aren't needed.

"Doing shows with everything from symphonic events to heavy metal, we haven't even had *one* component failure."

Cutter has likened the design process of his new sound system to the assembling of a vocal choir in a bandshell. "If you are very careful about who sings what parts, and where the different sections are seated, and have the right acoustical shell, then you'll have beautiful music that carries a great distance — it all works together." The Sundown Sound system has been pleasantly surprising many listeners this year. When the modular cabinets are assembled into a large array, a powerful, compact speaker system is created that seems to have fulfilled its design requirements (Figure 12).

## AUDIO TECHNIQUES, INC.

Audio Techniques, Inc., of Calabasas, CA, is a four year-old touring sound rental firm. Formed in 1981 by partners David Beecham and Bob Ludwig, the company has been supplying national touring systems for use in arenas by such entertainers as Chicago and John Denver. "David and I both have experience with major sound companies in the past, such as Clair Bros. and Stanal," explains Ludwig. "While working with these larger firms, we were able to gain insight into the many different problems that can be encountered when one attempts to haul a large sound system around the country."

In 1984, the partners decided to invest in a newly-designed sound sys-

**Figure 12: When a group of SDS enclosures are stacked together, a tight, compact array is formed. Here, one side of the 36-cabinet system supplied to the 1985 Del Mar Fair in Southern California is shown.**

**Figure 13: Audio Techniques' VLF enclosure is tuned to 27 Hz, and houses a pair of JBL 2245 loudspeakers; a three-way box also is shown.**

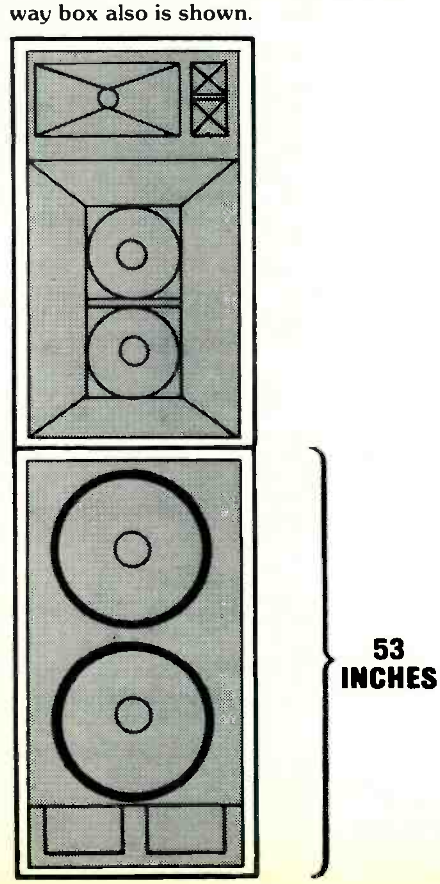

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Figure 14: Enclosures are attractively finished in a natural woodgrain. Canvas covers are used to protect the boxes in transit.

tem in order to enter the competitive touring arena-sound field. Rather than rely solely on a "seat-of-the-pants" approach to system design, Ludwig and Beecham felt that it was crucial for a *new* system to be assembled by making full use of the popularly-accepted, classical electro-acoustical design theory.

Mark Engebretson, a California-based audio consultant, was retained to work with the partners in the development of a two-box modular system. Engebretson, known for his former work with such firms as Altec-Lansing, Paramount Pictures and AB Systems, helped to steer Ludwig and Beecham toward a direct-radiating bass enclosure concept (Figure 13).

"They had certain preconceptions when we went into this project," Engebretson recalls. "They thought they wanted an all horn-loaded system, and I think that idea was based on the amount of capital that was available to build a rather large system. I was able to show them that, with proper venting and enclosure dimensions, getting some *real* low-end would not be a problem. We have a 27 Hz tuning for the VLF [Very Low Frequency] enclosure, as a matter of fact."

### Three-way Enclosure

The mid-bass horn characteristics desired for the Audio Techniques systems determined the basic size of the modular boxes. After allowing room for a 60- by 40-degree (JBL Model 2385) or 90- by 40-degree (Model 2380) Bi-Radial1™) horn and a pair of JBL Model 2404 compression tweeters, the cabinet dimensions came out to be 53 by 29 by 30.5 inches (H×W×D). The two-inch compression driver is a JBL Model 2445J.

The mid-bass chamber houses two

Figure 15: Audio Techniques relies on Yamaha mixing consoles. Shown here: a PM-2000-32, an M1516, and an M-508.

JBL Model 2202 12-inch loudspeakers, and has a coverage pattern of approximately 60 by 60 degrees over

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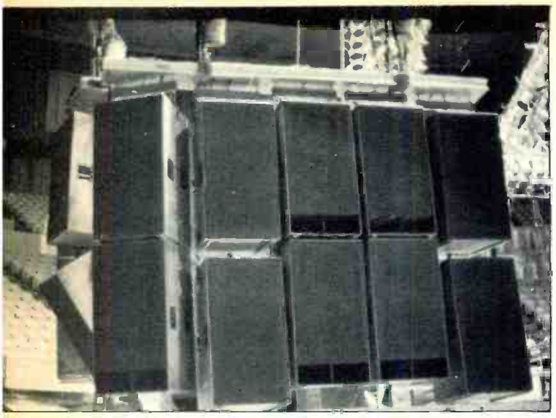

Figure 21: Cabinets are suspended from aluminum I-beams to form a quickly-assembled flying system.

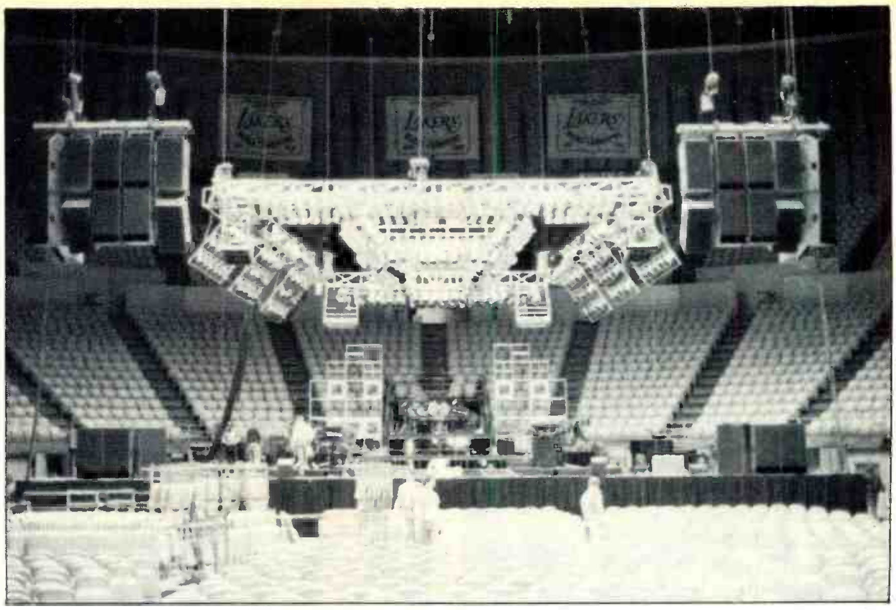

Figure 22: For Chicago's 1985 tour, Audio Techniques supplied a total of 40 cabinets, shown here at the Los Angeles Forum.



tom monitor systems, stage hardware, and equipment sales are all part of what we do. The touring sound systems are our primary focus, though."

Audio Techniques, cabinets have recessed Aeroquip fittings built into the back and side of each box. The assembly of easy-to-fly hanging arrays is possible with the use of nylon straps and aluminum I-beams. A typical 40-cabinet arena system comprises 22 VLF cabinets and 18 three-way boxes, providing a total of 44 18-inch speakers, 36 12-inch speakers, 18 two-inch compression drivers and 36 compression tweeters (Figure 21).

I heard this system in use at the Los Angeles Forum with Chicago. Its most notable characteristic, in this writer's opinion, was a very strong low-end response. The unique wood-grain finish offered a pleasing visual appearance. The time it took the stage hands to deal with the canvas covers seemed to be offset by the extreme ease with which the cabinets were flown by clipping the hanging straps into Aeroquip fittings (Figure 22).

### Systems Comparison

These two loudspeaker systems

take the same modular approach when attempting to fulfill the design requirements specified by two different sound system rental firms that are serving the same general market. While Audio Techniques' system has been designed with hanging arrays in mind, Sundown Sound's rig is primarily used in a large-stack format at stage level.

Both systems are powered by essentially the same wattage, as shown in Table 1. The same number of cone loudspeakers and compression drivers are combined into approximately the same cubic volume of cabinetry, although the Sundown Sound system would appear to offer somewhat smaller arrays due to the trapezoidal enclosures. (The firm is currently researching custom hardware for the development of hanging systems.)

The Audio Techniques system is a contemporary example of the approach taken by most major sound companies for portable, roadworthy hanging systems, while the Sundown System has taken advantage of such trendsetting techniques as trapezoidal cabinetry and time-corrected crossover circuitry.

While no two touring sound systems that are surveyed will *ever* match up detail for detail, it is significant to note that here are two different portable systems, both designed and built within the past year by unrelated, competitive firms with the help of nationally-recognized electro-acoustical consultants. Each loudspeaker system is closely matched in overall design format. Sundown's CDI system is perhaps more innovative.

Touring sound company personnel often hope for the ability to make a side-by-side comparison of such systems in the presence of measurement devices and system designers, so that an on-the-spot consensus can be reached spot regarding various characteristics of each system. Such events very rarely take place, due in part to logistical considerations. It would definitely appear, however, that "two-box" systems have come into their own, and these two growing rental companies have helped to define a new standard for custom-built touring concert sound systems. ■■■

**Companies mentioned in this article:**

Audio Techniques, Inc.
(David Beecham or Ludwig)
5169 N. Douglas Fir, Suite 4
Calabasas, CA 91302
(818) 992-6223

Sundown Sound, Inc.
(David Cutter)
P.O. Box 3085
Portland, OR 97208
(503) 230-0675

**Table 1: Comparison of Transducers, Crossover Frequencies and Power Ratings for the Two Sound Systems Discussed in this Article**

| Company | Frequency Passband: LOW | | MID | | HIGH | | SUPER-HIGH | |
|---|---|---|---|---|---|---|---|---|
| | Speaker | Power | Speaker | Power | Speaker | Power | Speaker | Power |
| **Sundown Sound:** | JBL 2245H 18-inch | 440W 4 ohms per pair | TAD TM1201 12-inch | 440W 4 ohms per pair | TAD 4001 | 240W 4 ohms per four | — | — |
| | (28 to 200 Hz) | | (200 to 1.6 kHz) | | (1.6 kHz and up) | | | |
| **Audio Techniques:** | JBL 2245H 18-inch | 450W 4 ohms per pair | JBL 2202H 12-inch | 450W 4 ohms per pair | JBL 2445J | 450W 4 ohms per four | JBL 2404 | 450W 4 ohms per four |
| | (27 to 220 Hz) | | (220 Hz to 1 kHz) | | (1 to 7 kHz) | | (7 kHz and up) | |

# DIGITAL SYNTHESIS AT PARKFIELD

## Emphasizing the Human Element While Producing Electronic Compositions for Television and Film Scores

*by Adrian Zarin*

As the electronic music scoring business grows by leaps and bounds — fuelled by the creative potentialities of digital control and multisystem interface —it's easy to lose sight of the "human dimension" that lies at the heart of every successful film or television score. But veteran film/TV composers John Parker and Alan Oldfield are one team that attempts to place the human element foremost in creating music with electronic instruments. Together, Parker and Oldfield make up a film scoring company known as Parkfield; the duo's headquarters is a synthesizer studio constructed in the garage of Oldfield's North Hollywood, CA, home. The equipment inventory is extensive, but Parker and Oldfield tend to list their own personal rapport as their *most* important, creative asset.

Parker describes the pair's professional relationship as an "ideal" partnership. "We first met sitting in with a jazz band in a little two-bit honky-tonk," he recounts. "I was bowled over by Alan's piano playing, but there was more to our meeting than that. In jazz, there's either an *instant* affinity between musicians, or there isn't. So much relies on instinct and improvisation; and you can tell right away whether a collaboration is working or not. I immediately sensed that I had a great rapport with Alan, both musically and philosophically; so I suggested a partnership."

Both partners have had long and varied careers in composition for the visual media. Parker started out as a jazz trumpeter, but was serving as musical director of *The Arthur Godfrey Show* by the early Sixties. In 1966, he relocated from New York to L.A., and began amassing a list of film and TV scoring credits that includes *S.W.A.T., Hawaii Five-O, Barnaby Jones, Streets of San Francisco, How The West Was Won, Dallas,* and *CHiPS*, among others. He soon found himself working with such top Hollywood studio synthesists as Clark Spangler and Ian Underwood, while his use of synths in the underscore for a 1970 *Gunsmoke* episode won him the Western Heritage Award for the best Western music of that year. For the past six years now, Parker has been creating all the music for the long-running series *Trapper John, M.D.*

In contrast with Parker, Alan Oldfield hails from more of a traditional, academic musical background—something which is perhaps in keeping with his role as Parkfield's synthesizer expert. After taking his BA and MA in music composition — at San Diego State University and UCLA, respectively — Oldfield went on to study in Paris with the prestigious composition instructor, Nadia Boulanger. After that, he took his Ph.D in music composition from North Texas State University. It was at that institution, during the late Sixties, that he had his introduction to synthesis via the school's prototype Moog Modular System. A subsequent teaching post at Southern Illinois University enabled him to work at another electronic music lab.

By the time Oldfield relocated to Los Angeles in 1975, to concentrate on film/TV scoring and session work, he had a formidable command of electronic instruments. Such a varied background has stood him good stead not only in the visual media but also on record dates, where he's done conducting, arranging and playing for Helen Reddy, Thelma Houston, Barry Manilow, Rick James and others.

Because Parker and Oldfield both have substantial "pre-electronic" musical backgrounds, their approach to composition and scoring often combines electronic techniques with elements of traditional "pencil and paper" orchestration.

All photography by Robin Zarin

### In-House Production Studio

Centerpiece of the Parkfield facility is a New England Digital Synclavier II digital synthesizer system. Presently, the system includes the basic Synclavier II keyboard/programming module, a VT-100 computer terminal, and NED's dedicated hard-copy music printer, which comes in very handy when Parker and Oldfield create scores for large orchestral sessions. Data storage and operation functions for the Synclavier are handled via a floppy disk drive and a Winchester hard disk.

Oldfield says he had several reasons for choosing the Synclavier from among the various high-end digital synthesizers currently on the market. "For one," he begins, "it's very much designed around players. Another reason why we got into Synclavier is that the company will be updating from this point on with software. Their approach is to build a complete music-composition system. They don't even call the Synclavier a synthesizer; it's *more* than that."

Two major updates that Oldfield is particularly eager to add to Parkfield's Synclavier system are poly-


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phonic sampling and MIDI interface capabilities, both of which will help centralize work around the unit. At the present moment, however, both of these functions are adequately covered by other equipment to be found in Parkfield's studio.

"We have access to digital sampling with our [E-mu Systems] Emulator II, " Oldfield explains. "And, because we have multitrack [tape] recording capabilities, our need for polyphonic sampling isn't that urgent."

In creating samples with the Emulator II, Oldfield likes to start by printing the sound on tape. He generally uses a single Neumann U87 mike to capture the sound to be recorded on Parkfield's Tascam 85B one-inch 16-track, or Technics Model 1500 two-track. From there, the audio is transferred to the Emulator's digital memory.

"I like to do it that way because it's more controllable," he comments. "In order to get the perfect sample, you often have to resample the sound several times until you've got *precisely* what you want. But, sometimes, you just can't repeat the sound *exactly* the same way; and you can really wear out the person who is producing the sound! So it's great to put the sound on tape first.

"The trade-off is that you probably lose a little bit of quality. But I've found that if you record on good tape, and use a good microphone, the loss of quality is negligable, and you still end up with a good sample."

### Visual Inspiration for Sound Composition

Although they have a substantial library of Emulator samples, whenever they can Parker and Oldfield prefer to create custom sounds for their compositions. More often than not, the inspiration for a particular sample will be a visual image — even when the team is not creating music specifically for a film or television show. After years of working in these media, Parker has evolved a uniquely visual approach to all composition. As an example of how this approach affects the team's use of sound sampling, he cites a recently-completed abstract composition titled *Pilgrims' Treasure*.

"For one particular passage," Parker recalls, "I envisioned a Shofar, a ram's horn, blowing across a canyon with a column of hooded figures crossing a mountain bridge. I felt that the only way to realize that would be to find a primitive horn sound that we could sample. As a trumpet player, I

**Parkfield principals John Parker (*left*) and Alan Oldfield in their 16-track facility.**

felt it would be appropriate for me to produce the sound. The only question was what to use for an instrument. We started out with a rolled-up copy of the *Hollywood Reporter;* I blew through it, but it didn't sound very good. So we tried a rolled-up copy of *Variety*, but that didn't work either.

"Then Alan remembered that there was a length of metal pipe in his backyard that had been left behind by somebody repairing the sprinkler system. It was about 10 foot long, about three-quarter of an inch in diameter —just perfect.

"After much effort, I finally produced one pure A-flat that could be sampled. We created a scale from that, which we used to play the melody I had written for that part."

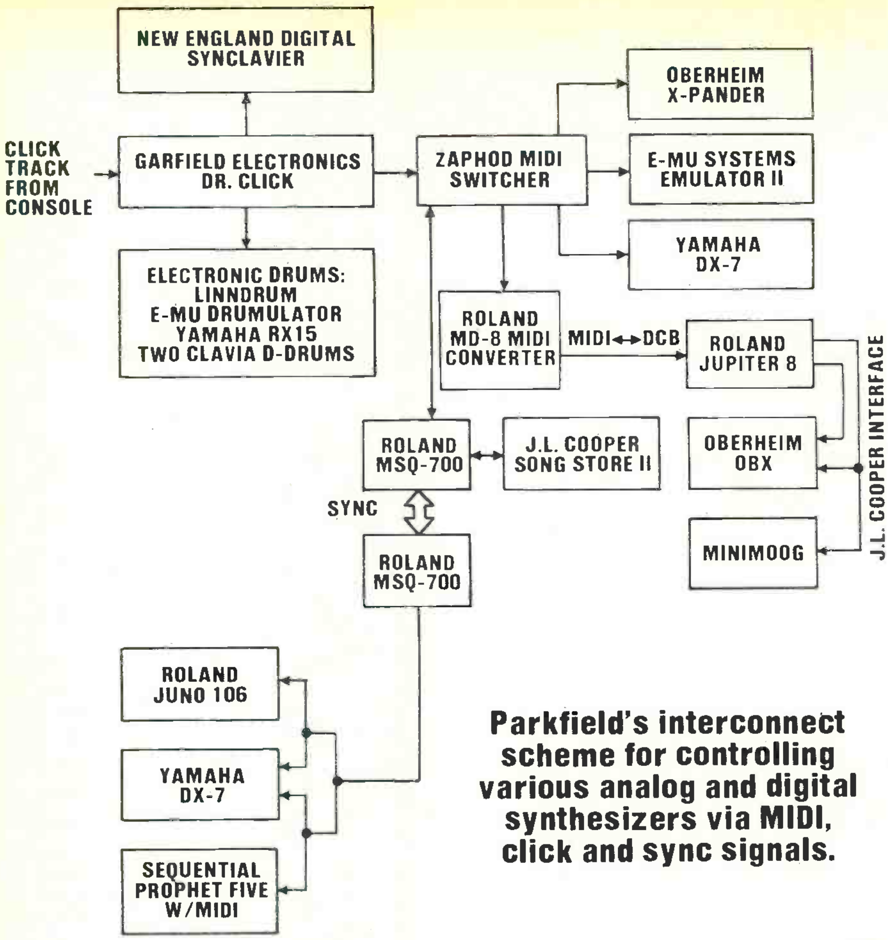
Along with the NED Synclavier II and E-mu Emulator II, Parkfield is stocked with a variety of other synthesizers and keyboard instruments, including a Yamaha DX-7, Roland Jupiter 8, Oberheim Xpander, a modified Oberheim OBX, and a Minimoog. Drum sounds are provided by a Linn-Drum, E-mu Drumulator, Yamaha RX-15, and two Clavia D-drums. These latter devices were manufactured originally in France; the design was later bought by E-mu Systems and manufactured as E-Drums. Both the French originals and their American counterparts are pressure-sensitive percussion pads that command one or two digital percussion sounds, loaded from memory.

The machines listed above form the core of Oldfield's travelling session set-up. It also includes two effects racks containing an Ibanez HD100 digital delay and UIE-405 Multi-Effects, and MXR 01Q digital reverb, a Boss DM-3000 delay, and a DOD R830 graphic equalizer. Oldfield also has two Master Touch breath controllers that he uses on his OBX and Jupiter synthesizers. His keyboard mixer is a RAMSA WR8112, which serves as a submixer to Parkfield's modified 16-channel TEAC Model 15 console. When working on a composition, Oldfield and Parker generally monitor on a pair of modified JBL component systems. For mixing however, they mainly rely on a pair of Yamaha NS10M "close-field" monitors. The studio's monitors are powered by a combination of Crown D150A and Yamaha P2050 amplifiers.

### Interface Techniques

On large projects, Parkfield's synthesizer ranks are swelled by Parker's personal set-up, which consists of a DX-7, Roland Juno 106, and a Sequential Prophet Five that has been modified to handle MIDI commands. In all, there's quite a large selection of synthesizers at Parkfield, which is somewhat unusual in electronic studios equipped with a powerful digital music system such as the NED Synclavier. But each machine, Oldfield feels, has its own use in the composition and recording processes. "I think all the different synthesizers have their own personalities, " he remarks. "So we like to use everything we have. Although the Synclavier can probably get all the sounds that we have on the other machines, we find it's actually easier to have six different syn-

**Centerpiece of the Parkfield studio is an NED Synclavier II synthesizer with dedicated music printer and hard-disk storage.**




**Parkfield's interconnect scheme for controlling various analog and digital synthesizers via MIDI, click and sync signals.**



thesizers if you're going to combine six different timbres to create a single sound. To us, that somehow seems easier than combining six different timbres in the Synclavier."

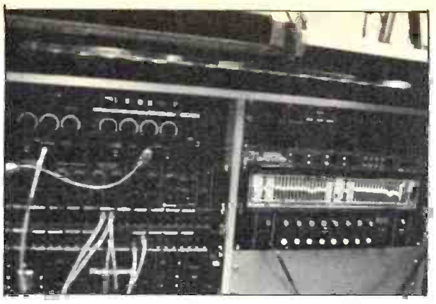
Of course, this kind of set-up means that all these synthesizers have to be interfaced and made into a single system. With the advent of a MIDI interface for the Synclavier, that machine's 32-track internal sequencer will be able to handle all sequencing and synchronization functions for every synthesizer in Parkfield's studio. At present, however, the Synclavier itself is the only instrument under the control of its own dedicated sequencer, the other machines being controlled by a pair of Roland MSQ-700 MIDI sequencers. One MSQ-700 controls Parker's three synthesizers, while the other drives all of Oldfield's machines, and is connected to a J.L. Cooper Electronics Song Store II, an external disk drive that increases the -700's data-storage capacity.

A Garfield Electronics Doctor Click is used to synchronize the two MSQ-700's to the Synclavier's sequencer, and to lock the entire system to sync code or a click track recorded on the multitrack master tape. Oldfield's non-MIDI synths — the OBX, Minimoog and Jupiter 8 — enter the system by means of a custom J.L. Cooper interface unit.

"Very few were ever made," says Oldfield of the Cooper interface unit. "It's sort of a pre-MIDI interface. There was a switching device put on every key of the OBX, which makes it possible to control any other instrument by means of an RS-232 [serial] interface."

At present, Oldfield has the OBX connected to the Minimoog and Jupiter 8 which, in turn, is connected to a Roland MD-8 MIDI converter that translates the Jupiter's DCB (Digital Communication Bus) interface into MIDI commands. All of which brings the Jupiter and the instruments connectedto it, under MIDI control. As an alternative sequencing system for these instruments, the OBX and the Jupiter are each connected to their own dedicated JLC-quencer, a single-channel, polyphonic sequencer manufactured by J.L. Cooper Electronics. These sequencers operate on a 24 pulse-per-quarter-note clock, and can be synchronized with the other systems via a Garfield Doctor Click.

Working with various MIDI configurations over the past few years, Parker has experienced some of the dreaded time-lag problems encountered by synthesists who use the serial MIDI interface to connect together a



*Left:* **Outboard units include a Roland DM-3000 delay, Ibanez UIE-405 and HD-100 effects, MXR 01 reverb and DOD R830 graphic;** *Below:* **Tascam Model 15 mixer;** *Right:* **Tascam MS-16 16-track.**

large number of machines. Most of Parkfield's problems, however, have been brought under control through the use of a Zaphod MIDI switcher. "I send my master signal [from the Doctor Click] to the Zaphod," Parker explains; "and that sends a signal to all the MIDI instruments at the same time. This way, you don't get a serial buildup; so I've noticed very little time lag on our projects.

"There have been some occasions where we've had different amounts of time lag among certain instruments, so we sometimes have to use a digital delay to compensate for that time difference. Basically, though, it hasn't been a major problem for us."

"I don't know if there will be time-lag problems when we go to the Synclavier MIDI port, but I do know that it should make a big difference in the way we operate. We'll be able to sit down at the Synclavier and record all our ideas on its 32-track recorder. Then we'll be able to have individual tracks control our other synthesizers —the DX Jupiter, or whatever. This ability to run everything from one central keyboard will be a new breakthrough in our control building."

### Compositional Approaches

Over the past year, Parker and Oldfield have evolved an approach to composing which, they feel, is somewhat different from the way in which other film and TV composers collaborate.

Oldfield explains: "If you look at a lot of the top TV writing teams —Mike Post and Pete Carpenter, for example — you'll see that each member of the team basically works alone. One person writes the first, third and fifth themes, say; and the other writes the second and fourth.

"What John and I have tried to do instead is to merge our compositional abilities on each piece, rather than working on separate pieces. While that has taken many different forms, essentially one of us will come up with a basic element that is recorded or at least worked out in a rough form of some sort. Then, the other one takes off at that point, modifying that original element and adding to it.

"We always find that, by allowing ourselves to let the other person's influence into the picture, we wind up with a different direction than either of us would have taken by ourselves."

The basic musical elements that form the starting point for Parkfield compositions are often written out on paper first. From there, the composers



begin to draw on the resources of multitrack recording — both on analog tape, and by using digital sequencers — to bring the composition to completion. This technology provides them with a medium for exchanging musical ideas, developing them together and instantaneously hearing the results played back.

"Technically," Oldfield elaborates, "there is a kind of general approach that we take. We tend to start from the rhythm, and then work from the bottom up. Rhythmically, if there's a drum part involved that's where we usually start. We lay out the rhythmic activity in sequenced patterns; I like to lay that down for the simple reason that everything else has to be played to that rhythmic foundation.

"Then, after all those sequential

*Left:* **Parkfield's extensive analog and digital keyboard collection;** *Below:* **closeup of primary rack with Oberheim Xpander, E-mu Systems Emulator II and Yamaha DX-7;** *Right:* **secondary rack housing a Minimoog, Roland Jupiter 8, Oberheim OB-X and RAMSA WR8112 mixer.**

elements are down, I generally like to go to the bass concept — although we've done it every imaginable other way as well. From the bass, we usually go to a general center sound; then we add the melody. And finally, we add all the little enhancing elements — the special effects."

To establish the basic tempo for a piece of film or TV music, Oldfield likes to rely on the traditional "Click Book" that has been a staple of scoring for years. Using it, he selects a tempo that suits the visual material, and which becomes the basis for a physical click track used to drive Parkfield's various synthesizers via

the Doctor Click. Or, it is expressed as a tempo that is entered into one of the drum machines.

At present, Parkfield has no means of locking the sequencers or tape machines to a work print of the picture. The studio is equipped, however, with a JVC VHS half-inch videocassette deck and monitor. The composers periodically use the VCR to check their score against a VHS copy of the work print, and thereby gauge how well picture and music are meshing.

Up until now, the composers have been more-or-less confined to fixed tempos when scoring with sequencers and drum machines. But that era will end, they announce, with their next planned acquisition: Oracle composition software for Parkfield's Commodore 64 computer.

Oldfield explains what new capabilities the Oracle software will provide: "One of the limitations of music recorded to click has been what they call ametric tempos — 7/4 going into 3/4, and then to a 4/8 bar. The reason why it has been a problem is that you can't vary the tempo — the rate of the click itself. With Oracle, that will change.

"What it does is divide every second into 1,000 portions, so you can have minute control over time. It's like a super metronome, controllable and variable. With it, we can generate a variable click to drive our sequencers."

### *TRAPPER JOHN, M.D.*: From Parkfield to Scoring Stage

A good example of the Parkfield approach to technology and television music is provided by one recent project: an updating and re-recording of the main title music that Parker wrote for *Trapper John, M.D.* The new arrangement was worked out at Parkfield, where Oldfield and Parker recorded what the latter refers to as a "template" for the final version, which was recorded at 20th Century Fox. His metaphor is an apt one, when you consider that the drum-machine data created for the Parkfield recording furnished the guiding structure for the final recording.

"We knew the exact amount of time we had," Parker explains: "1 minute and 14.7 seconds. We knew the click we wanted and we knew the tempo. This enabled us to build the [Linn] drum track at Parkfield. We downloaded the program to cassette and gave it to Joe Pocaro, who is my percussionist. He loaded it into his drum machine, the new [E-mu Systems] Drumulator SD-12. For the session, he embellished the program with his drum kit, adding statements and little things that were needed as punctuations. The result was fantastic. And it's the first time I'm aware of that a combination of drum machine and real drums was used on the main title for a TV series."

The idea of orchestrating music with a combination of acoustic and electronic instruments is one that Parker and Oldfield find particularly exciting. In addition to its blend of acoustic and electronic percussion timbres, the new *Trapper John* title also combines the sound of a 37-piece orchestra with the work of two synthesists, Alan Oldfield and Tom Ranier.

"A lot of people talk about electronic devices 'threatening the livelihood of musicians'," Parker comments. "But I feel that, instead of threatening their livelihood, electronic instruments are *stimulating* the livelihood of musicians by encouraging people to use acoustic instruments in ways they've never been used before.

"I think that the *Trapper John* scoring session illustrates that brilliantly. The first phrase of the main title is played by synthesizers alone: it's Alan playing a sound that we created partially at Parkfield, and partially at Fox by combining four or five synthesizers with MIDI. Then, for the second phrase, I brought in French horns and trombones, playing the melody in unison. But I also programmed a sound on my [Yamaha] DX-7 for Tom Ranier to play along with those instruments. It was a very beefed-up clavinet type of sound, with lots of attack.

"And between the brass instruments and the electronic sound, I think we produced something the *transcended* any of the sounds taken by themselves."

Whether they're combining orchestral and electronic timbres or converting click-book timings into drum machine data, John Parker and Alan Oldfield make one very important point through their work. They illustrate admirably that the traditional approach to creating film and TV music is by no means at odds with the electronic age. In taking the best elements of both traditional and electronic methodologies, they've established a firm basis for their own collaborative relationship as composers. By all indications, it's a relationship that will continue to evolve. ■■■

# PERFORMANCE ASSESSMENTS OF STUDIO MICROPHONES

## Continuing Stereo Evaluations at the University of Iowa School of Music

*by Lowell Cross, Professor of Music and Director of Recording Studios*

The previous articles in this series, published in the April 1984, December 1984, and February 1985 issues of *R-e/p*, were concerned primarily with contemporary microphones that are readily available to studio engineers. Only three "vintage" examples were evaluated: the Neumann SM69 and U47 vacuum-tube condenser, and the RCA 77-DX ribbon. Because of the wide-spread interest in older designs, particularly tube condenser models, the editors of *R-e/p* received suggestions that we should investigate more of the "legendary" microphones from the past.

This prospect was sufficient to arouse the interest of Bill Bradley, a Chicago-based collector and dealer, who was persuaded to bring many of his highly prized items to Iowa City for our May 13, 1985 session. His cooperation, and that of the importers of AKG, Coles, Milab, Sanken, and Schoeps models, provided us with the rare privilege of comparing venerable and justly-renowned tube units to present-day condenser and ribbon microphones.

### The Recording Session

Every effort was made to duplicate the concert hall setup and conditions of the previous experimental sessions. Soprano Carol Meyer and pianist Patricia Cahalan graciously performed the Mozart and Gershwin songs in Clapp Recital Hall as before. However, the presence of so many intriguing microphones, classic and contemporary, prompted us to record other musical combinations under exactly the same circumstances. I am indebted to my faculty and student colleagues at The University of Iowa School of Music for their willingness to become involved in our project. The music which they performed in front of 16 types of microphones (all in stereo) during our May 13 session follows:

Mozart: "Come My Love" from *The Impresario*.

Gershwin: "Summertime" from *Porgy and Bess*; Carol Meyer and Patricia Cahalan (Figure 1).

Beethoven: Sonata No. 9 in A for violin and piano ("Kreutzer"), op. 47; first movement; Professors Leopold LaFosse and Kenneth Amada (Figure 2).

Beethoven: Sonata No. 21 in C ("Waldstein"), op. 53; first movement; Professor Amada (Solo piano: Figure 1 or 2).

J.S. Bach: "Contrapunctus IX" from *The Art of the Fugue*, arranged by John Glasel for brass quintet; student ensemble (Figure 3).

All photography by the University of Iowa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## STEREO MICROPHONES
*Professor Cross continues his Subjective Evaluations*

The review of the Sanken CU-41 microphone published in the October 1985 issue of *R-e/p* mentioned an important change that has been made in our Control Studio A monitoring equipment: the JBL Model 4320s have been replaced with tri-amplified Klein+Hummel O92 standard reference loudspeakers. Without detracting from the excellent long-term performance of the JBL monitors, I can report that this replacement has greatly benefited our microphone evaluations and other recent studio projects. The effortless transparency of the O92s has allowed a more revealing aural "view" of the behavior of the test microphones than was available previously[1]. Otherwise, all studio equipment and techniques have remained the same as those described in the earlier articles in this series, including the use of AKG K-141 stereo headphones to supplement the monitor loudspeakers.

The practice of using a pair of Neumann TLM170s and an SM69 (tube) stereo microphone has been maintained as a control measure. The Neumann U47 tube units from the Fifties were evaluated again, so that they could be directly compared to other studio models from that period. These Neumann microphones belong to our Recording Studios; all others were loaned by Bill Bradley or the various importers, as we have already noted.

Table 1 lists specifications for the current group of 16 models, and contains information regarding the temperature, barometric and humidity conditions of the May 13 session. Since 16 stereo pairs cannot be recorded simultaneously on a 24-channel tape, two takes were required for each musical selection. For the second take, certain microphones were retained with no change ("The Tube," VIP-50, TLM170, CMC541U) while others were retained with a pattern change (cardioid to hypercardioid — C12, C12A, C414 EB/P48, M49). The second take therefore was used to record different microphones *and* to make comparisons between cardioid and hypercardioid patterns where feasible. Note that the necessity of recording two takes per selection has prevented an ordering of the microphones in exact alphabetical and numerical sequence.

With one exception (see below), no low-frequency rolloff or capsule attenuation switches were activated; no microphones were modified for this recording session; and no procedures nor placements were used that could be considered improper by the manufacturers. All were condenser models, except the Coles 4038 ribbons, whose polar patterns are figure-of-eight only (bidirectional). They were set up in the classic Blumlein configuration of a coincident XY pair; the left and right microphones were each angled 45 degrees away from the sound sources.

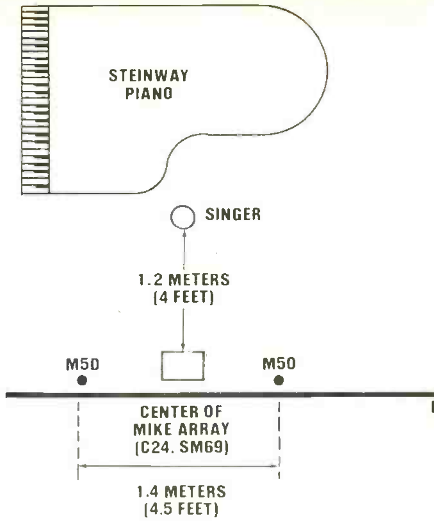


**Figure 1: Layout for piano and vocal sessions.**

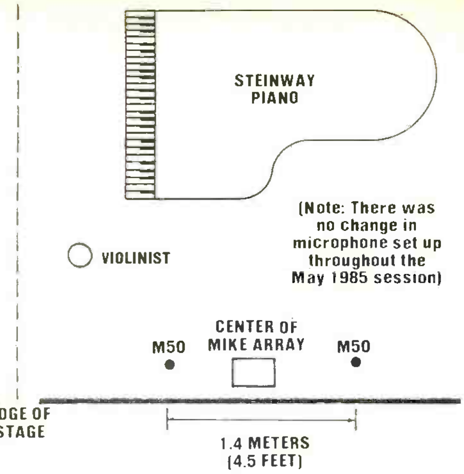


**Figure 2: Layout for piano and violinist sessions**

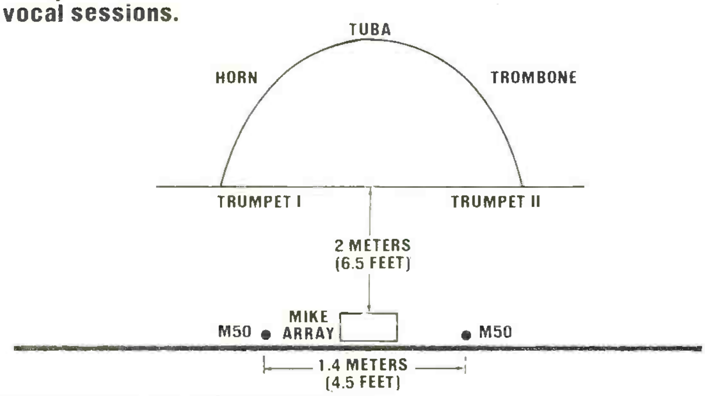


**Figure 3: Layout for brass quintet sessions.**

The two tube stereo microphones (C24 and SM69) were recorded in the MS technique via Neumann Z140 matrix transformers. The directional condenser units, either cardioid or hypercardioid, were slightly spaced or as "near-coincident" as the mounting arrangement would allow. (It should be evident that it is impossible to locate 30 microphones — two stereo and 14 pairs — so that all receive *exactly* the same acoustical information.) The only omnis in the collection — the Neumann M50s — were spaced 4½ feet (1.4m) apart; see accompanying photographs and draw-ings of the setup in the concert hall. No electret models were included in this evaluation.

### The Microphones

• The **AKG C12** is one of the most highly regarded large-diaphragm studio models in microphone history. Introduced in 1953, it offered nine remotely-controlled patterns: omnidirectional, cardioid, and figure-of-eight, plus six additional intermediate settings, including hypercardioid. It was imported into North America from the Fifties until production ceased in 1964[2]. The C12 made use of one-half of a 6072 dual-triode tube (essentially a premium version of the common 12AX7 or ECC83); the 6072 also is found in the new AKG Tube (see below). This famous Austrian design was related to the Telefunken ELA M250 and ELA M251, manufactured by AKG in the Fifties, when Telefunken and Neumann were ending their association. Quite recently, C12s gained international attention as vocal microphones in the *We Are The World* sessions. The C12 was a true Viennese classic: a beautifully crafted design capable of a wide range of applications — and capable of commanding a price of over $2,000 each in today's used equipment market.

• One might assume from its model number that the **AKG C12A** was a slightly revised version of the C12, but it was actually a very different microphone. The C12A, the tube counterpart and predecessor of the contemporary C414EB models (which use field-effect transistors), was manufactured from 1965 until 1972. The physical appearance of the C12As and the C414EBs is the same except for minor details; both types use large capsules (one-inch diameter) directly descended from those found in the earlier C12, C24, and ELA M250/251 models. The C12A, like the C12, offered a selection among nine remotely-controlled polar patterns, but differed from the latter in utilizing a 7586 nuvistor instead of the 6072 dual triode, and in offering two low-frequency rolloff settings (-7 and -12 dB at 50 Hz), in addition to a "flat" position.

• The C414EB was evaluated in the two-part December 1984/February 1985 report; the **AKG C414EB/P48** is the most recent offering in this well-known line of studio microphones. As noted above, its tube ancestor was the

## TABLE 1: SPECIFICATIONS OF 16 MICROPHONES USED IN STEREO EVALUATION SESSION

Recording session of May 13, 1985 (class project for Music 25:214, Recording Techniques, Spring Semester 1985). Concert Hall temperature: 68°F (20°C). ±3°F. Relative humidity: between 50 and 60%. Barometric pressure: 29.86 inches of mercury (758.5 mmHg, 101.1 kPa).

**TAKE #1**

| | Brand and Model Numbers | Serial Numbers | Date of Manufacture | Nominal Source Z | Recommended Load Z | Approximate Console Gain | Patterns |
|---|---|---|---|---|---|---|---|
| 1. | AKG C12 (tube) | 1943 1954 | early Sixties | 200 ohms | 1,000 ohms nominal | 45 dB | cardioid |
| 2. | AKG C12A (tube) | 2341 2963 | late Sixties | 200 ohms | 500 ohms or above | 55 dB | cardioid |
| 3. | AKG C414EB/P48 | 25825 26015 | 1985 | 200 ohms | 500 ohms or above | 45 dB | cardioid |
| 4. | AKG C460/ CK61 | 07992-0374* 07682-1026* | 1985 | 120 ohms | 500 ohms or above | 45 dB | cardioid |
| 6. | The AKG Tube | 230 529 | 1985 | 200 ohms | 1,000 ohms or above | 40 dB | cardioid |
| 7. | Milab VIP-50 | 10970 10972 | 1985 | 200 ohms** | 1,000 ohms nominal | 45 dB | cardioid |
| 8. | Neumann U47 (tube) | 1637 2990 | 1953 1956 | 200 ohms | 1,000 ohms or above | 30 dB | cardioid |
| 10. | Neumann M49 (tube) | 686 1182 | 1954 1956 | 200 ohms | 1,000 ohms or above | 35 dB | cardioid |
| 11. | Neumann M50 (tube) | 025 558 | 1950 1956 or 1957 | 200 ohms | 1,000 ohms or above | 30 dB | omni-directional |
| 13. | Neumann TLM170 | 3/100 4/100 | 1983 | 150 ohms** | 1,000 ohms minimum | 40 dB | cardioid |
| 14. | Schoeps (tube) M221B/M934B | 013 3099 | 1958 or 1959 1968 or 1969 | 200 ohms | 1,000 ohms nominal | 40 dB | cardioid |
| 16. | Schoeps CMC541U | 9226-31106* 9870-31108* | 1985 | 35 ohms** (actual) | 600 ohms or above | 35 dB | hypercardioid |

**TAKE #2**

| | Brand and Model Numbers | Serial Numbers | Date of Manufacture | Nominal Source Z | Recommended Load Z | Approximate Console Gain | Patterns |
|---|---|---|---|---|---|---|---|
| 1. | AKG C12 (tube) | 1943 1954 | early Sixties | 200 ohms | 1,000 ohms nominal | 45 dB | hypercardioid |
| 2. | AKG C12A (tube) | 2341 2963 | late Sixties | 200 ohms | 500 ohms or above | 55 dB | hypercardioid |
| 3. | AKG C414EB/P48 | 25825 26015 | 1985 | 200 ohms | 500 ohms or above | 45 dB | hypercardioid |
| 5. | AKG C24 (tube) | 532 | late Sixties | 200 ohms | 1,000 ohms nominal | 45 dB | mid:cardioid side:bidirectional |
| 6. | The AKG Tube | 230 529 | 1985 | 200 ohms | 1,000 ohms or above | 40 dB | cardioid |
| 7. | Milab VIP-50 | 10970 10972 | 1985 | 200 ohms** | 1,000 ohms nominal | 45 dB | cardioid |
| 9. | Coles 4038 (ribbon) | — — | 1985 | 300 ohms | over 1,000 ohms | 60 dB | bidirectional |
| 10. | Neumann M49 (tube) | 686 1182 | 1954 1956 | 200 ohms | 1,000 ohms or above | 35 dB | hypercardioid |
| 12. | Neumann SM69 (tube) | 1061 | 1968 | 200 ohms | 1,000 ohms minimum | 40 dB | mid:cardioid side:bidirectional |
| 13. | Neumann TLM170 | 3/100 4/100 | 1983 | 150 ohms** | 1,000 ohms or above | 40 dB | cardioid |
| 15. | Sanken CU-41 | 1081 1082 | 1985 | 150 ohms | 600 ohms or above | 45 dB | cardioid |
| 16. | Schoeps CMC541U | 9226-31106* 9870-31108* | 1985 | 35 ohms** (actual) | 600 ohms or above | 35 dB | hypercardioid |

***Amplifier serial number — capsule serial number; **Transformerless**
**Z = impedance.**
**Nominal microphone input impedance of Neve 5315/24 console: 1.2 kohms (transformer; balanced and floating).**

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C12A, which it closely resembles in physical appearance. The principal difference between the P48 version and the other C414s is the requirement of a 48-volt phantom power supply. (Earlier models needed only 12 volts; higher voltages were stepped down internally.) The change, which is also a welcome step toward standardization, has resulted in an improvement in the self-noise specification of the microphone. The 12-volt C414EB is still available.

C414s offer four polar patterns — the usual three, plus hypercardioid — as well as a full complement of capsule attenuation and LF rolloff capabilities. AKG claimed in 1984 that the C414EB/P48 has "the highest dynamic range of any microphone we know[3]." Its versatility, specifications, attractive yet unobtrusive appearance, have made it one of the most popular large-capsule studio microphones.

Stage area of Clapp Recital Hall during May, 1985, evaluation sessions, showing microphone layout and distance to grand piano.

• The **AKG C460/CK61** is also an upgraded version of a microphone that we discussed in an earlier report. The new CK61 cardioid capsule is described as offering a more linear frequency response than the familiar CK1, which is still available. The C460 amplifier module operates on 48V phantom powering, offers the option of 0 or -10 dB capsule attenuation, and provides "flat," 70 Hz, and 150 Hz bass rolloff settings. The evolution of these small (18mm diameter) "combo" microphones extends back through the popular C451/C452EB series, to the C60 and C61 models of the tube era.





• The **AKG C24** was the stereo version of the C12. It was sold in North America from 1964 until as recently as 1977/8, when it was replaced by the current C422 (field-effect transistor) stereo microphone. The C24 contained two, one-inch capsules of the C12-type, one of which could be axially rotated. The coincident XY and MS (Lauridsen) intensity-stereo techniques were the intended applications of this system, which offered nine remotely-controlled polar patterns per capsule.

• The continuing strong interest in tube condenser designs has led to the recent introduction of **The AKG Tube**. In response to this demand, AKG has invoked memories of the C12. The Tube comes with the same 6072 dual triode, a one-inch, double-diaphragm capsule, and nine polar patterns, all put together in a somewhat "nostalgic" looking cylindrical package. However, the new design provides the additional refinements of 0, -10, and -20 dB capsule attenuation, two LF rolloff characteristics plus a "flat" setting, and the option of increasing the amplifier's output level by 10 dB. The increase in output is accomplished by switching in the second triode section of the 6072 tube. (Only one half of the 6072 was used in the C12.) The capsule is the same as the one used in the C414 series. The Tube is an expensive product ($1,700), yet sales have been brisk.

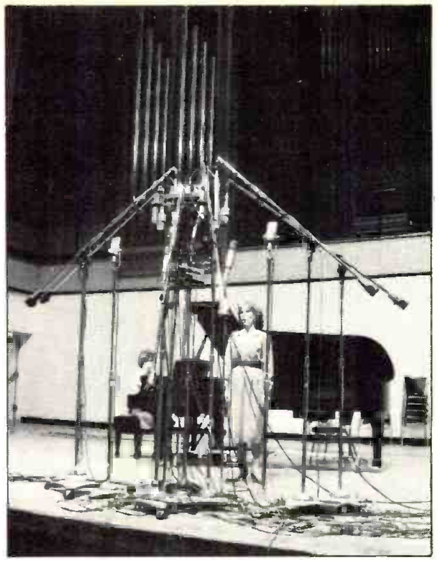
• In my October 1985 *R-e/p* review, the **Milab VIP-50** was compared to the Sanken CU-41, AKG C414EB/P48, and Neumann TLM170. This Swedish import is characterized by its rectangular capsule; five patterns (omni, wide-angle cardioid, cardioid, hypercardioid, and bidirectional); transformerless circuitry; and the option of either a line- or microphone-level output. At over $1,500, the VIP-50 is the second most expensive of the current models under discussion, after The AKG Tube.

• The famous **Neumann U47** tube model was introduced in 1948. Georg Neumann (1898-1976) began his company 20 years earlier in Berlin, and by 1930 was manufacturing the CMV-3 "bottle," marketed by Telefunken as the SO-16. The CMV-3 (or SO-16) and subsequent CMV-3a microphones had plug-in heads, which provided the three most common polar patterns: bidirectional (CM-7 head), cardioid (CM-8), and omni (CM-9). These products figured prominently in German broadcasting throughout the Thirties and during World War II. Neumann's marketing association with Telefunken (a subsidiary of the two large German electrical manufacturers, AEG and Siemens) lasted until 1958, and accounted for the "Tele" trademark used on the U47 and other microphones. After the end of the war, Neumann began to rebuild and retool his Berlin factory; the U47 was his first postwar product[4].

The original U47 made use of a Telefunken VF-14 pentode tube, wired as a triode, and offered the engineer a choice between omnidirectional and directional (supercardioid) patterns. Alas, the VF-14 is no longer available. Neumann today sells the U47fet model, which retains the same capsule (now supercardioid only) but with a solid-state amplifier design. Approximately 5,000 U47 tube microphones were manufactured through August 1960. They are now collector's items, each worth up to $2,000 or more in the used microphone market[5].

• The **Coles 4038** ribbon microphone has evolved over a long period of development by the British Broadcasting Corporation. The natural pickup pattern of a ribbon (open on both sides) is bidirectional, or figure-of-eight; this is the single polar characteristic available from the 4038. (The RCA 77-DX and other types have offered the selection of multiple patterns by varying mechanical elements coupled to the ribbon.) The Coles 4038 offers no LF rolloff capability or other options.

• The **Neumann M49** and **M50** were introduced soon after the U47; there is an approximate correspondence between the model numbers of these three microphones, and their dates of inception. The M49 and M50 were developed in collaboration with a design team led by Dr. Herbert Grosskopf of the German Institute for



**Piano and vocalist session with featured artist Carol Meyer.**

Broadcasting Technology (Institut fuer Rundfunktechnik, or IRT). Dr. Grosskopf's work, an extension of the pioneering efforts of von Braunmuehl and Weber in the Thirties, resulted in fundamental patents that formed the basis of subsequent variable-pattern condenser designs. The M49 utilized Grosskopf's 1949 patent covering electrically variable condenser capsules, which could be remotely controlled. (His 1952 design for mechanically controlled capsules was licensed to Schoeps.) The 28mm (1.1-inch) capsule of the M49 incorporated gold-impregnated polyvinyl chloride (PVC) membranes; polyester foil (or Mylar, a DuPont trademark) has replaced PVC in the capsule technology of the present[6].

The M49 was a large, imposing microphone. Its protective outer mesh screen had a characteristic shape similar to that found on the more recent, and smaller U67, U87, U89, and TLM170. The design of the capsule backplate in today's U89 and TLM170 still maintains close similarities to the original M49 concept[7]. These two contemporary units share the capability of the M49 to generate at least five patterns: omni, wide-angle cardioid, hypercardioid, and figure-of-eight.

The M50's physical appearance was identical to that of the M49, but there the similarity ended. The M50 was basically an omnidirectional design, but with very special characteristics. The engineering goal for the microphone was one of constructing a pressure transducer with a polar pattern which became *directional* at high frequencies, while exhibiting a rising free-field (i.e., close range or non-reverberant) high-frequency response.

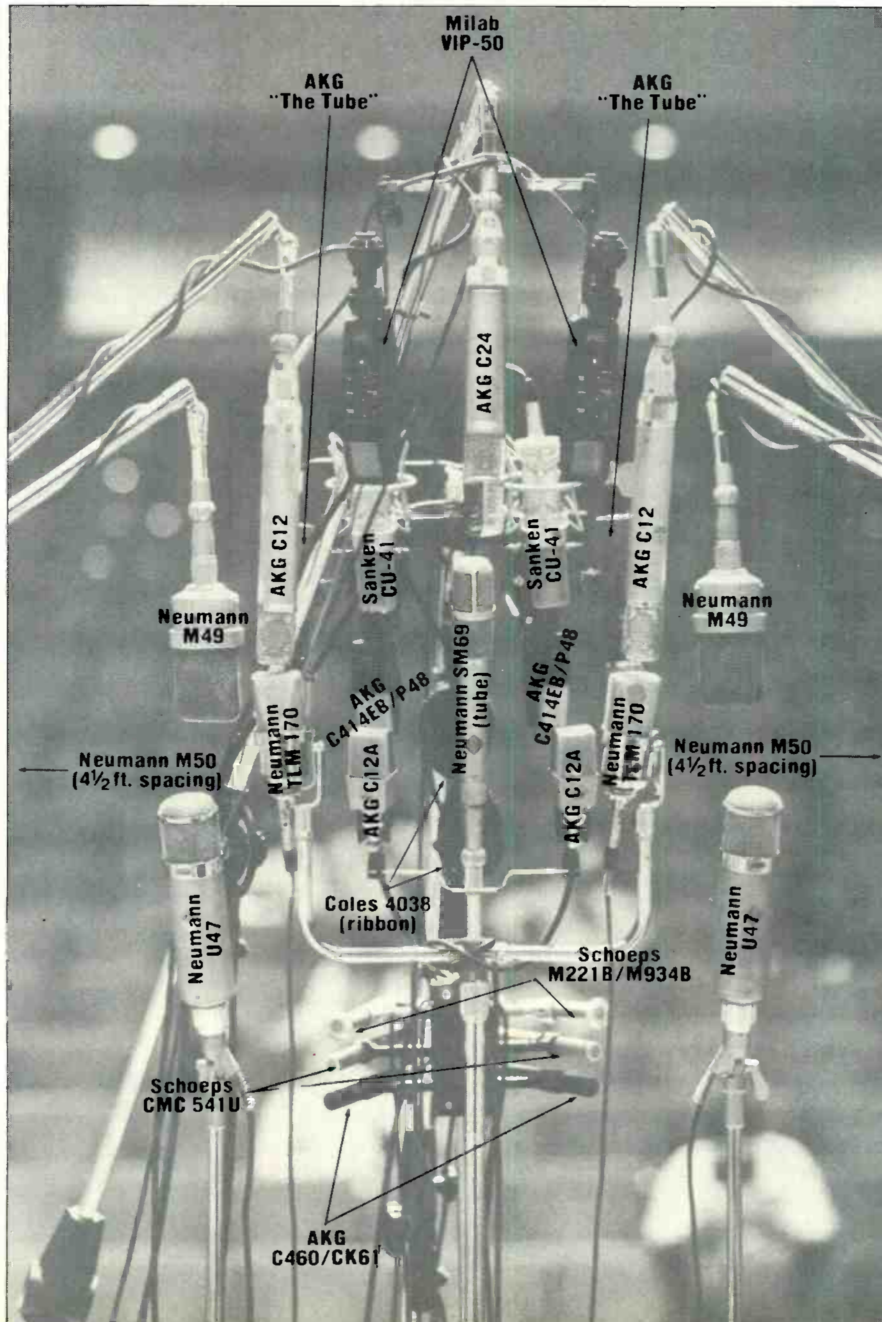
## STEREO MICROPHONES
*Professor Cross continues his Subjective Evaluations*

In combination, these properties yielded a microphone with an omnidirectional pattern below 1 kHz (becoming more and more unidirectional above 2 kHz), yet with uniform high-frequency response in the diffuse, or reverberant, field.

Such characteristics made the M50 ideal for single-microphone *monophonic* recording and broadcasting of large musical groups in reverberant concert halls and radio studios — a typical requirement of European broadcasting in the Fifties. The Grosskopf-Neumann collaborators accomplished their goal for the M50 by imbedding a tightly stretched, gold impregnated, 12mm (half-inch) polystyrol diaphragm on the surface of a plastic sphere 40mm (1.6 inches) in diameter; the result was one of the most unusual condenser microphone designs to date. (For a period of its history, nickel was impregnated on the diaphragms; the last of the M50 capsules were again manufactured with the gold formulation.)

Both the M49 and M50 employed a subminiature Telefunken AC701K triode, which had been adopted as the standard tube for condenser microphone electronics in German broadcasting. Only a very limited quantity of AC701k tubes are still available in 1985. And speaking of limited quan-

Close-up detail of microphone array for evaluation sessions.



tities — M49s and M50s are extremely scarce and very expensive in today's "vintage" microphone market.

• The **Neumann SM69** tube stereo microphone, a direct competitor to the AKG C24, was manufactured from the mid-Sixties until about 1970. The SM69 was the immediate predecessor of the present solid-state models, the SM69fet and USM69, and it in turn was preceded by the first Neumann stereo microphone, the SM2. Manufactured from 1956 into the mid-Sixties, the SM2 contained nickel-membrane capsules of the same type found in the earliest "KM" series of miniature tube microphones.

Beginning with the SM2, all Neumann stereo systems have had rotating upper sections and a selection of nine patterns per capsule, to accommodate the coincident XY and MS (Lauridsen) stereo techniques. The SM2 and SM69 models utilized two each of the ubiquitous AC701k tubes, as one would expect with microphones having to meet the requirements of German broadcasting.

• **Neumann TLM170**: the control microphones; see page 123 of the April 1984 issue of *R-e/p* for a full description.

• The **Schoeps M221B/M934B** was a miniature dual-pattern microphone also using the AC701k tube, manufactured from the late-Fifties until the end of the Sixties. (The first part of the model number is the amplifier designation; the second denotes the capsule.) The change between omni and cardioid patterns was accomplished by mechanical means — the 1952 Grosskopf design mentioned above — including a special plastic ring that screwed onto the end of the capsule. The ring formed part of the acoustical delay network used to obtain the cardioid pattern[8]. The shape, diameter (20mm or 0.8-inch), and other dimensions of the M221B/M934B are very similar to those of the solid-state microphones manufactured by Schoeps today.

Adherents to tube-microphone technology will be interested in the announcement by the Schoeps factory that enough parts remain to manufacture 20 final M221B/M934C microphones. When released, these will be identical to our review microphones, except for the absence of the screw-on capsule ring — hence the "C" designation. Further information about this very special offering can be obtained from Posthorn Recordings, New York, at (212) 242-3737.

• The **Sanken CU-41**, reviewed in the October issue of *R-e/p*, is distinctive in its use of two capsules mounted one above the other, which combine electrically and form a cardioid-only pattern. The design of this Japanese manufacturer takes advantage of the higher-output, lower-noise performance of a large diaphragm over most of the audio range, and the use of a smaller one to minimize high-frequency resonances and "colorations." With its single pattern, absence of options for equalization or capsule attenuation, and impressive price ($1,495), the CU-41 is a "purist's" microphone.

• The **Schoeps CMC541U** is identical to the CMC54U discussed in the December 1984/February 1985 report, except that it has a hypercardioid rather than a cardioid capsule. It is a member of the Colette modular family, well known for an exceptionally comprehensive range of capsules and options, high-output signal levels, and transformerless circuitry. Interest in the use of hypercardioid patterns for "minimalist" stereo recording has prompted the inclusion of these microphones in our evaluation. The Schoeps MK41 capsule is the only interchangeable one known to the author to obtain the hypercardioid pattern from a fixed, single-pattern diaphragm.

### The Hypercardioid Pattern

The hypercardioid is approximately halfway between cardioid and bidirectional (figure-of-eight). A highly significant feature is its property of extracting the *least* amount of sound power from the diffuse field (i.e., the least amount of room reverberation) of all of the common types[9]. Of course, the smaller lobe of the pattern faces the rear, which must be considered when positioning the microphone. (See Figure 4 for a comparison of the hypercardioid and supercardioid patterns.)

An omnidirectional microphone is considered to have a "random energy efficiency" of 100%; the figure for a hypercardioid is only 25% (bidirectional and cardioid, each 33 1/3%; supercardioid, 27%)[10]. In comparison to omnis, hypercardioids pick up only one-fourth as much room reverberation, and they may be used in diffuse-field applications up to twice the distance from the sound source without sacrificing clarity or creating feedback: important considerations when maximum isolation and separation are required. These characteristics have contributed to the present interest in using hypercardioids for stereo recording.

### Tubes versus Transistors

We referred to this provocative topic earlier in our series of evaluations, without attempting any resolution of the issue. In fact, the issue may not

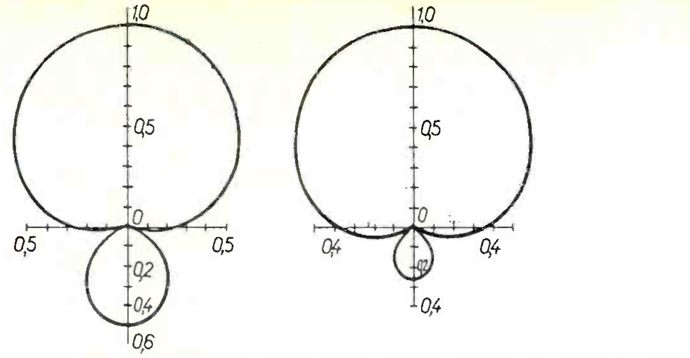



**Figure 4: Hypercardioid *(left)* and supercardioid *(right)* patterns. (Taken from Reference #9, used with permission.)**

need to be resolved, because (fortunately) there are many excellent examples of both types.

Before the advent of the transistor, condenser microphone design depended exclusively upon tube electronics. The vacuum tube, with its extremely high input impedance, was a "natural" component for matching condenser capsules to the remainder of the program chain. Field-effect transistors (or FETs, also extremely high input impedence devices), have now completely replaced tubes as impedence converters in condenser circuitry, with the noteworthy exception of the recently-introduced specialist unit—The AKG Tube. Another important transistor design not requiring FETs is based on the RF modulator-demodulator principle, originally developed by Stefan Kudelski (of Nagra fame) and marketed by Sennheiser[11].

No other topic in present-day audio mythology, with the possible exception of the "Digital versus Analog" debate, excites as much passion as the matter of vacuum-tube electronics. Yet the "Tube versus Transistor" issue is hardly a debate at all when it comes to condenser microphones. The truly loyal (and vocal) disciples are those in the tube camp: their microphones have achieved mythical, legendary, and unquestioned status; their venerable items can bring resale prices of over $2,000 each; they use their microphones to make up-to-the-minute digital recordings without any concern about the anomalous use of old equipment in a high-tech situation — all because tubes are simply supposed to "sound better" than transistors. The FET defenders are rarely, if ever, as visible or as vocal as the tube proponents.

Tubes and transistors *are* fundamentally different. In tubes, electrons are boiled off a hot filament-cathode combination, sent past "valve"-like control elements called grids, and attracted to a positively-polarized anode. All of this activity takes place in a near-vacuum, usually enclosed in a relatively large glass envelope. Transistors, field-effect and otherwise, are solid-state devices. The emitter-base-collector electrodes (source-gate-drain in FETs) accomplish their control or amplifying functions within a microscopically small region of a DC-biased crystalline semiconductor such as silicon. Even though the functions of the cathode, grid(s), and anode are analogous to those of the emitter, base, and collector (or source, gate and drain in FETs), the basic physical properties of the two technologies are not at all comparible. It is no wonder that tubes and transistors are *expected* to "sound different."

Under certain circumstances, they do sound different. In one of the few scientific investigations into the subject, Gotham Audio's Russ Hamm has noted that a tube and a transistor will exhibit quite dissimilar distortion characteristics as each begins to approach its respective overload point. His overall conclusion is worth repeating: "The basic cause of the difference in tube and transistor sound is the weighting of the harmonic distortion components in the amplifier's overload region[12]." Most of the time, microphone electronics of either type are not pushed to that point.

Other factors that can influence the sound of vintage units are transformers (all tube microphones have them); aging of the capsules and other components (especially PVC diaphragms); and the effects from tubes which are in themselves "microphonic." It has been reported that the VF-14s in U47s may contribute sonic information of their own by resonating sympathetically at high sound-pressure levels[13]. (Most people would consider "microphonics" to be a defect.) Transistors, on the other hand, are virtually immune from generating audio signals when subjected to vibration.

It remains the goal of good audio design to produce electronic circuitry that is as distortion-free, wide-band,

and noise-free as possible. That goal has been accomplished quite elegantly with tubes *and* solid-state devices — note the great success of both types. (It is not a formidable engineering challenge to design relatively simple impedance converters of either category that satisfy such criteria. Capsule design is another matter.) I assert that the very transparent circuits found in high-quality tube *and* transistor microphones permit us to listen back to the behavior or the capsules themselves. My assertion has been reinforced by recent conversations with engineers and equipment designers whose work I respect: Jerry Bruck and Joerg Wuttke (Posthorn Recordings and Schoeps), Russell Hamm (Gotham Audio Corporation), Stephen Julstrom (Shure Brothers, Inc.), Derek Pilkington (AKG Acoustics, Inc.) and David Muller (The University of Iowa).

So I am prepared to make an unequivocal statement: the highly prized sonic qualities of tube condenser microphones are principally the result of the capsule designs found in those microphones, and *not* the tubes. Those older designs produce colorations *that can provide forms of enhancement*. Newer capsules usually have less coloration — by deliberate intent on the part of the manufacturers — and therefore are less "interesting" to many engineers for that very reason.

### Microphone Evaluations

This method of evaluating microphones involves subjective choices. As before, I acknowledge the influence of personal, non-empirical factors in the process, on my own part and on the part of our other participants. However, I believe that any problems of subjectivity have been more than offset by the practical, real-world conditions of our recording sessions. We were privileged to listeny (IRT, mentioned above) and the German Tonmeister Association (Verband Deutscher Tonmeister, VDT) are involved in very similar microphone listening tests has contributed to my confidence in our evaluation technique.

We now present the composite ratings of the 16 microphones from the fourth and most recent session, averaged from responses received from 25 listeners, and shown in Table 2. During both recording and playback, we deliberately avoided all factors that could be considered advantageous or disadvantageous to any of the microphones. In the audition sessions, the microphones were identified by numbers and heard in no particular order, with as much skipping around and returning to requested models as possible. Our listeners were asked to evaluate the microphones on a scale of 1 (least favored) to 10 (most favored); it was possible for them to assign the same ranking to one or more units if they wished.

**Table 2: Composite Ratings for Evaluation #4, May 13, 1985**

| | |
|---|---|
| Neumann M50 (tube) | 7.94 |
| Neumann TLM170 (FET) | 7.79 |
| Neumann SM69 (stereo, tube) | 7.46 |
| Sanken CU-41 (FET) | 7.21 |
| Neumann M49 (tube) | 7.16 |
| AKG C12 (tube) | 7.09 |
| AKG C12A (tube) | 7.01 |
| Schoeps CMC541U (FET) | 6.98 |
| The AKG Tube | 6.90 |
| AKG C414EB/P48 (FET) | 6.86 |
| Coles 4038 (ribbon) | 6.82 |
| AKG C460/CK61 (FET) | 6.75 |
| Neumann U47 (tube) | 6.74 |
| AKG C24 (stereo, tube) | 6.68 |
| Schoeps M221B/M934B (tube) | 6.66 |
| Milab VIP-50 (FET) | 6.55 |

In the previous sessions, the control microphones (Neumann TLM170s) received the following composite scores:

| | |
|---|---|
| Evaluation #1, Dec. 1983 | 7.44 |
| Evaluation #2, May 1984 | 7.62 |
| Evaluation #3, June 1984 | 7.31 |

Of course, one must take a cautious

approach to the interpretation of these rankings. A significant factor is the influence of *stereo* quality on the perception of *microphone* quality. Our listeners consistently have given high ratings to the microphones that conveyed to them a sense of spaciousness and hall ambience. Their reactions are entirely appropriate for the musical repertoire chosen for our sessions. Furthermore, the majority of our listeners — music students and faculty — are performers themselves. They tend to prefer reproduced sound which *heightens*, and not just *documents* (no matter how accurately), the musical performance. Professors LaFosse (violinist) and Amada (pianist) independently singled out the tube SM69 as a first choice — without any knowledge of which microphone they were choosing. Both liked the rendering of the hall ambience as heard through the MS technique; both liked the rather bright, "present" quality for which the SM69 is known. Performers are accustomed to hearing music "up close" without the high-frequency attenuation that audiences experience in concert halls.

Another factor concerning microphone technique was the preference of most of our listeners for a cardioid pattern over a hypercardioid — which simply indicates to me that the hypercardioid microphones such as the Schoeps CMC541U were fulfilling their design criteria. Less extraction of the sound power from the diffuse field equals less hall ambience, less spaciousness, and less "atmosphere." The music we chose was well suited for Clapp Recital Hall; the minimizing of its acoustical qualities was not appropriate for these performances. However, if recordings must be made in an overly reverberant hall, microphones with hypercardioid patterns are well worth considering.

Table 3 lists the interpolated composite "rankings" of 40 microphones from the four evaluation sessions. A total of 75 listener responses have been tabulated to date. In the case of models that were recorded on more than one occasion (SM69fet, SM69 tube, TLM170, U47 tube), the scores have been averaged.

### Personal Evaluations

Without any further elaboration of my awareness of the subjective elements involved here, I will proceed to offer my own opinions of the microphones under review. The *Neumann TLM170* remains at the top of my list with a 9+, but I am also very partial to its close relative, the *U89*, as well as the *Schoeps MSTC54* and *CMC54U*, the *Calrec MkIV Soundfield*, and the *Neumann KU81* artificial head — reviewed in the earlier articles. The Ambisonic Soundfield technique is one of the most important design concepts in recent microphone history. I

**TABLE 3: Subjective Rankings of All 40 Microphones from Four Evaluation Sessions**

| Rank | Microphones | Stereo Technique | Composite Score | Evaluation Number |
|---|---|---|---|---|
| 1. | Calrec MkIV Soundfield | Stereo output, similar to matrixed MS | 8.02 | 2 |
| 2. | Neumann M50 (tube) | Widely-spaced omni (4.5 feet) | 7.94 | 4 |
| 3. | Neumann SM69fet | MS, matrixed | 7.81 | 1, 2 averaged |
| 4. | Schoeps MTSC54 | ORTF (near-coincident) | 7.58 | 2 |
| 5. | Neumann KU81 | Binaural dummy head | 7.56 | 2 |
| 6. | Neumann TLM170 | Slightly-spaced cardioid | 7.54 | 1 thru 4 averaged |
| 7. | Neumann SM69 (tube) | MS, matrixed | 7.51 | 3, 4 averaged |
| 8. | Neumann U89 | Slightly-spaced cardioid | 7.38 | 1 |
| 9. | Schoeps CMC54U | XY coincident cardioid | 7.34 | 2 |
| 10. | AKG C34 | MS, matrixed | 7.22 | 2 |
| 11. | Sanken CU-41 | Slightly-spaced cardioid | 7.21 | 4 |
| 12. | Neumann M49 (tube) | Slightly-spaced cardioid | 7.16 | 4 |
| 13. | Neumann U87 | Slightly-spaced cardioid | 7.13 | 1 |
| 14. | Milab XY-82 | XY coincident cardioid | 7.12 | 3 |
| 15. | AKG C12 (tube) | Slightly-spaced cardioid | 7.09 | 4 |
| 16. | Neumann U47 (tube) | Slightly-spaced cardioid | 7.06 | 1, 4 averaged |
| 17. | AKG C12A (tube) | Slightly-spaced cardioid | 7.01 | 4 |
| 18. | Schoeps CMC541U | Slightly-spaced hypercardioid | 6.98 | 4 |
| 19. | Milab LC-25 | Slightly-spaced cardioid | 6.91 | 3 |
| 20. | The AKG Tube | Slightly-spaced cardioid | 6.90 | 4 |
| 21. | AKG C414EB/P48 | Slightly-spaced cardioid | 6.86 | 4 |
| 22. | Milab DC-96B | Slightly-spaced cardioid | 6.86 | 3 |
| 23. | Coles 4038 (ribbon) | Blumlein (XY coincident fig. 8) | 6.82 | 4 |
| 24. | Neumann KMF4 | XY coincident cardioid | 6.80 | 2 |
| 25. | AKG C414EB | Slightly-spaced cardioid | 6.75 | 2 |
| 26. | AKG C460/CK61 | Slightly-spaced cardioid | 6.75 | 4 |
| 27. | Neumann KM84 | Slightly-spaced cardioid | 6.72 | 1 |
| 28. | AKG C24 (tube) | MS, matrixed | 6.68 | 4 |
| 29. | Schoeps M221B/M934B (tube) | Slightly-spaced cardioid | 6.66 | 4 |
| 30. | AKG 460B | XY coincident cardioid | 6.59 | 2 |
| 31. | Crown PZM-31S | PZMs™, on floor (spacing of 4 feet) | 6.57 | 3 |
| 32. | Milab VIP-50 | Slightly-spaced cardioid | 6.55 | 4 |
| 33. | AKG C452EB | Slightly-spaced cardioid | 6.54 | 1 |
| 34. | Neumann KM86 | Slightly-spaced cardioid | 6.50 | 1 |
| 35. | Bruel & Kjaer 4007 | Widely-spaced omni (9 feet) | 6.43 | 2 |
| 36. | Shure SM81 | Slightly-spaced cardioid | 6.38 | 1 |
| 37. | Calrec CM2050C | XY coincident cardioid | 6.04 | 2 |
| 38. | Electro-Voice RE15 | Slightly-spaced cardioid | 5.75 | 1 |
| 39. | RCA 77-DX (ribbon) | Slightly-spaced cardioid | 5.38 | 1 |
| 40. | Sennheiser MKH405 | Slightly-spaced cardioid | 4.27 | 1 |

hope that Professor Stanley P. Lipshitz will prove to be correct in his prophecy that "This is the way of the future[14]." I am also prepared to give high marks to the *Bruel & Kjaer 4000* series as individual units, but I do not think that they or other omnidirectional microphones are appropriate for stereo applications.

This brings us to the *Neumann M50*, a special omni design. It conveys a "wide open" and ingratiating reproduction of the original musical event (score: 9+). The M50's rich, warm low-end is exceptionally well balanced by a clean, transparent midrange and crisp, scintillating highs. (Am I describing a vintage wine or a vintage microphone?) The upper formant and sibilant properties of the singer's voice were sharply detailed by the M50, as were the harmonic content of the violin sound and the transient character of the piano. This wealth of detail is present along with a healthy measure of reverberation because of the fundamentally omnidirectional pattern.

The overall effect was one of actually *flattering* the musical combinations that we recorded. I attribute these qualities of the M50, which do offer a distinctive form of enhancement, to its unique capsule construction (described above) — but, unfortunately, no longer being manufactured. However, in spite of all of the foregoing praise, I must grant that there are very serious drawbacks to this intriguing microphone. As a pressure transducer, or omni, it cannot be recommended for stereo recording. The RTW Model 1260 stereo correlation meter recorded an unsatisfactory range of -0.6 to +0.7 for the M50 pair, spaced 4½ feet apart. And rather considerable problems for potential users are its extreme scarcity and present resale value: $2,000 and up, each.

I am almost as well disposed toward the *M49*, which *can* be used properly in stereo applications, with its full complement of directional patterns — another 9+ rating. The M49 has a rather "velvety" high-frequency characteristic, and a full, robust low-end. In this sense, its sonic properties can be compared to those of the TLM170 when the two are used as cardioids. This is another rare and very expensive microphone from the past.

Also ranking in the 9 to 9+ range are two more excellent contemporary FET designs, the *Schoeps CMC541U* and the *Sanken CU-41*. Previously, the Schoeps MSTC54 and the CMC54U (both cardioid) received higher composite scores from our listeners than the one given to the CMC541U (hypercardioid) in the most recent evaluation. Their preferences are attributable to the greater amount of hall reverberation heard from the cardioids. Otherwise, the CMC541U exemplifies the excellent qualities of Schoeps products: clean, uniform response characteristics, low self-noise, and high output levels from transformerless circuitry.

The Sanken CU-41 is an impressive new entry into the ranks of the elite. Its fine audio qualities result from an extended high-frequency response, excellent cardioid pattern characteristics, and wide dynamic range. The CU-41's only potential drawbacks are cost, the absence of user-adjustable features (such as pattern selection, LF equalization, or capsule attenuation), and slightly less low-frequency output than some of the other first-class microphones.

The two *Neumann SM69s* (tube and transistor), along with the oldest and newest large-diaphragm AKG models, the *C12*, the *C24*, and the *C414EB/P48*, come next in my rating scheme (8+). We have reviewed the SM69 models in previous articles; suffice it to say that I hear from them the high-frequency emphasis associated with "condenser sound." (When will Neumann release an "SM89" with TLM170/U89 capsules?) As already noted, the SM69 was the favored microphone of the members of our performing faculty who took part in this evaluation; they liked its bright qualities in combination with the MS stereo technique.

The classic C12 has a very positive appeal; I can understand its popularity for recording vocals. Its brightness in such applications is hardly a detriment to its overall performance, but rather can be used as a means for enhancement. The C24, the stereo version of the C12, conveys these same bright but appealing qualities. The "brightness peak" of the C12/C24 is in a slightly higher part of the spectrum than that of its famous competitor, the U47 — also very popular as a vocal microphone. Bill Bradley's C24 exhibited some noise problems during our session (all of his other microphones performed perfectly); perhaps this situation accounted for its slightly lower rating than the C12s by our group of listeners.

The C414EB/P48 is a very good large-capsule studio microphone, especially in view of its price. It does not seem to have as extended a high-frequency response as some of the other new models, and there is a certain amount of "condenser-like" emphasis in the 6 to 10 kHz region. However, it is versatile, attractive in appearance, and capable of excellent low-noise performance.

Next come two tube units, the

*Schoeps M221B/M934B* and the *Neumann U47*; each receives a rating of 8. Both offer switchable omni and cardioid patterns, but otherwise they are not at all alike. The Schoeps is small and unobtrusive, like its present-day FET descendants, while the Neumann is large and imposing. The former has a clean, crisp sound quality similar to contemporary Schoeps models, with some high-frequency emphasis. The U47's response peak is lower and more pronounced, giving it a very "personal" sound that works well when close, bright, "up-front" qualities are desired. Because of its special sound, the U47 commands a highly respected position in the audio world today. However, it will not appeal to engineers who are seeking accuracy.

The *Milab VIP-50* and *AKG C12A* are lower in the rating scheme than one would expect (7+ to 8–), simply because of their overly bright qualities. Unlike the SM69, C12, C24, U47, and M221B/M934B, the high-frequency emphasis of either the VIP-50 or the C12A is not particularly enhancing or ingratiating. Both tend toward a hard-edge condenser sound; I am sure than our listening panel gave the otherwise high-quality Milab a lower rating for that reason. The VIP-50 has the potential to rise to the top ranks of elitist microphones once its frequency response is made more uniform. (The two Milabs on loan for this evaluation were pre-production examples of a model still undergoing development; a stereo version may be released in the future.) The C12A simply has the bright sound that was identified with AKG line in past years; today's models have much more uniform response.

Our present discussion concludes with the *AKG C460/CK61*, *The AKG Tube*, and the *Coles 4038* ribbon. All three of these microphones could have received higher scores than my 6 to 6+ anking if they did not have certain problems that became evident during our evaluation. The C460/CK61s loaned to us by AKG were extremely sensitive to handling and air turbulence. When in the presence of even moderate air motion (from persons walking close by, etc.), they both exhibited low-frequency overload or "blocking." To give them every possible chance of a good rating, we activated their low-frequency rolloff switches at the 70 Hz position. This cured the problem, but the extreme low-end was obviously attenuated in comparison to the other microphones. (Derek Pilkington of AKG Acoustics, Inc. was surprised by this condition; but he did inform me that AKG has extended the low-end response of the C460 FET circuitry and the CK61 capsule below that of previous models such as the C451.) In other respects, this microphone has a good high-frequency response and low self-noise characteristics.

Once they knew what they were listening to, many of our participants described The AKG Tube as rather "dull." With well-designed electronics (tubes nonwithstanding), it should sound very much like the C414-EB/P48, whose capsule it shares. The answer to its dullness in our particular situation, with a 400-foot (120m) cable run between Clapp Recital Hall and our control rooms, may have been revealed in a review by Hugh Ford that appeared two years ago in *Studio Sound*: "Whilst not being the very quietest of microphones, in other respects 'The Tube' came out well but some care may be needed when using long cables in view of the fairly high output impedance at high frequencies[15]."

Even though our evaluation of The AKG Tube could not reveal what its high-frequency performance might be over a short cable run, I feel that I do not need additional information to suggest that there are much better values in the AKG line. The same capsule is available in the C414EB/P48 at less than half the price, and the FET unit is quieter and much easier to use, thanks to 48V powering. At $1,700, I feel that The Tube is just too expensive for its true level of performance. For only a few hundred dollars more, diehard tube enthusiasts might be able to buy the real thing, a

**Two Neumann M50 omnidirectional mikes were spaced 4.5 feet apart along axis of the microphone array.**

**Peter Nothnagle *(left)* and Bill Bradley with collection of tube-mike power supplies and microphone connect cabling.**

C12, and own a piece of microphone history.

The only microphone that did not work upon arrival was a Coles 4038 ribbon. After about half an hour of troubleshooting, we discovered that one of the special cables supplied with the 4038s was wired incorrectly (the shield and a signal lead were reversed). But my main concern with this model was its very low signal output, requiring approximately 60 dB of preamplifier gain the the console. Even with the admirably quiet Neve electronics, there is audibly more background hiss on the Coles tracks than on any of the others — including those of the old tube models. This ribbon unit is supplied with a nominal 300-ohm source impedence, which is a bit high (30 ohms on special order).

I did find the 4038 to have the soft, "warm" quality that is often admired in a ribbon microphone. The low frequencies and midrange were well balanced, but there was some rolloff at the high-end. (We could not take the time to determine if this was the result of the long cable run, or the microphone's natural high-frequency characteristics.) I much prefer the sound of the 4038 to that of our RCA 77-DX ribbons, which have many more operating features, but which to my ears have unpleasant resonances and a deficient high-frequency response.

Ribbons are favored by many engineers for recording brass instruments, especially in close situations. Since they do not have the extended HF response of many condenser units, they can act as equalizers as well as transducers. In this evaluation, I liked the 4038s more on the vocals than on the brass, which sounded too dull for my taste. Furthermore, the transient response of virtually all of the condenser models was audibly superior to the ribbons in the reproduction of the piano.

Our listeners did enjoy hearing the hall acoustics via the true Blumlein technique, as revealed by the bidirectional patterns of the Coles microphones. But mounting them in a coincident arrangement was no easy task — they are quite awkward and heavy. And, unfortunately, they are also physically unattractive in comparison to the elegant Austrian, German, and Japanese condenser microphones, both old and new.

### Conclusions

The participants in this and previous evaluations were fortunate to have heard music reproduced through some of the finest microphones ever made. Present-day models exemplify the ongoing accomplishments of contemporary technology, but the older ones reveal how truly advanced microphone design was in the Forties, Fifties and Sixties. I am as impressed by the remarkable qualities of "tube mikes" as anyone; I take special pride in our ownership of certain well-preserved examples from that bygone era.

But in praising tube microphones, we cannot lose sight of the basic underlying contributions of condenser *capsule* designers in advancing the state-of-the-art. We must continue to treasure and use those older condenser models with their highly desirable forms of coloration, which are unlikely to be duplicated as technology "advances."

I wish to express my gratitude to our audio engineer, Peter Nothnagle, for his devoted and invaluable assistance throughout all of our recording and playback sessions for this evaluation. ■■■

### Notes and References

1. The 240-watt, tri-amplified K+H O92 loudspeaker systems have an advertised free-field transmission characteristic of ±1.5 dB, 80 Hz to 12.5 kHz with third octave pink noise; ±3 dB over the same range with a sinewave signal; and less than 1% second- and third-order broadband harmonic distortion at 80 dB SPL (2m distance).
2. I am indebted to Derek Pilkington of AKG Acoustics, Inc. for his generous assistance in providing historical information about the C12, C12A, C24, and ELA M250/251 microphones.
3. See "Editorial note," *Journal of the Audio Engineering Society*, Vol. 32, No. 9 (September 1984), p. 678.
4. Winn Schwartau and David Smith, "Neumann: The History of Tube Condenser Microphones," *Recording Engineer/Producer*, Vol. 11, No. 1, (February 1980), p. 82.
5. I wish to thank Stephen F. Temmer, founder, and Russell O. Hamm, president, Gotham Audio Corporation, for their kind cooperation in providing extensive historical data on Neumann microphones.
6. F.W.O. Bauch, "New High-Grade Condenser Microphones," *Journal of the Audio Engineering Society*, Vol. 1, No. 3 (July 1953), pp. 232-240; reprinted from *Wireless World*, February 1953.
7. Stephen F. Temmer, private conversations with the author.
8. Valuable information about the history of Schoeps microphones has been graciously provided by Albert Grundy, importer of Schoeps products during the tube era; Jerry Bruck, Posthorn Recordings, New York; and Joerg Wuttke, Schalltechnik Dr.-Ing. Schoeps GmbH, Karlsruhe, Germany.
9. Gerhart Bore, *Microphones for professional and semi-professional applications* (transl. S.F. Temmer). Berlin: Georg Neumann GmbH, 1978. p. 19.
10. Michael Rettinger, *Practical Electroacoustics*. New York: Chemical Publishing Co., 1955, p. 42.
11. Schwartau and Smith, *op. cit.*, p. 90.
12. Russell O. Hamm, "Tubes vs. Transistors — Is There an Audible Difference?" *Journal of the Audio Engineering Society*, Vol. 21, No. 4 (May 1973), p. 272.
13. Schwartau and Smith, *op. cit.*, pp. 92-93.
14. Stanley P. Lipshitz, "Stereo Microphone Techniques: Are the Purists Wrong?" Audio Engineering Society Preprint No. 2261(D-5), May 1985, p. 38.
15. Hugh Ford, "Reviews: Microphones," *Studio Sound*, Vol. 25, No. 12 (December 1983), p. 72.

# STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

DALLAS SOUND — redesigned Studio C

AURA SONIC — ASL recording truck

GNOME SOUND — owner Bruce Nazarian

*Northeast:*

☐ **SOUND IDEAS** (New York City) has finished upgrading its Studio B by installing a 56-input **Solid State Logic SL6000 E** Stereo Video Music System with eight stereo modules, and a **Studer A-80 MkIV** 24-track machine. Monitors are provided by **UREI 813Bs**. *151 West 46th Street New York, NY 10036 (212) 869-2666.*

☐ **AURA SONIC** (Flushing, New York) has built an **ASL Mobile Audio/Video** recording truck specifically for pre- and post-production work for both audio and audio-for-video production. Featured in the vehicle is a 76 audio-mike line audio/video interface panel, four individual communications lines, eight video lines, 10 **Telco** lines, plus 816-point **Bantam**, and 24-point **BNC** video patch bays. Equipment spotlighted in the mobile is a **Sound Workshop Series 30** console, an **Ampex MM-1200** 24-track, two **Denon DR M4** cassette decks, two **Otari MX5050B II** machines, and an assortment of signal processing equipment. Control room monitors are **K&H 092** tri-amp powered speakers, **Yamaha NS-10s, EV Sentry 100s, Auratone 5Cs,** and **UREI 813Bs**. *140-02 Poplar Avenue, Flushing, NY 11355 (718) 886-6500.*

☐ **PRESENCE STUDIOS** (East Haven, Conneticut) is reported to be the only studio in the northeastern area to add a **Fairlight CMI Series III** 16-voice synthesizer to its instrument array. A **Yamaha DX-5,** a six-piece **Sonar** synthesized drum kit, and microphones from **Sennheiser** and **AKG** were also added to the studio's equipment list. *461 Main Street, East Haven, CT 06512 (203) 467-9038.*

☐ **UNIQUE RECORDING** (New York City) has purchased a **Fairlight II** synthesizer, a second **Linn 9000** drum synthesizer with sampling card and 128K RAM memory expansion card, three **Yamaha REV-7s** a third **AMS RMX-16** digital reverb, a second **Publison Infernal 90** compressor/limiter. *701 Seventh Street, New York, NY 10036.*

☐ **NATIONAL VIDEO CENTER/RECORDING STUDIOS** (New York City) have added a third audio-for-video studio to its complex. The 24-track room contains an automated 28-input automated **Sony MCI 500** console a **Dolby**-capable **JH-110** 24-track machine, and **Ampex MTR 100** eight-, four- and two-track machines, interlocked by a Quantec 3.10 Q.Lock synchronizing system. Monitoring is provided by **JBL/UREI Model 813Bs** powered by **Crown** amplifiers. Designer and co-founder of the facility **Irving Kaufman** says that the new studio is situated in close proximity to the other six National Recording video post-production suites. *460 West 42nd Street, New York, NY 10036 (212) 279-2000.*

☐ **MEDIASOUND** (New York City) has redesigned its new Studio C to include a new 32-input **Harrison Raven** console linked to a **Studer A80 MkII** 24-track, a pair of **A80 MkII** two-tracks, and a **Sony MCI JH-114** four-track machine. Studio A has acquired **Eventide 969 Harmonizer**, and **Linn 9000** drum machine with disk drive. In addition, facility president **Michael Hektoen** says that he has had great success with his new **MIDI Impact** room which provides the following equipment: 64-track MIDI/SMPTE sequencing/timecode software for **IBM, Macintosh,** and **Commodore** computers; a 32-track **NED Synclavier** digital music system with SMPTE timecode interface capabilities; sampled sound and percussion library; interface capabilities for MIDI-capable guitars; and an "array of phase distortion digital synthesizers." *311 West 57th Street, New York, NY 10019 (212) 765-4700.*

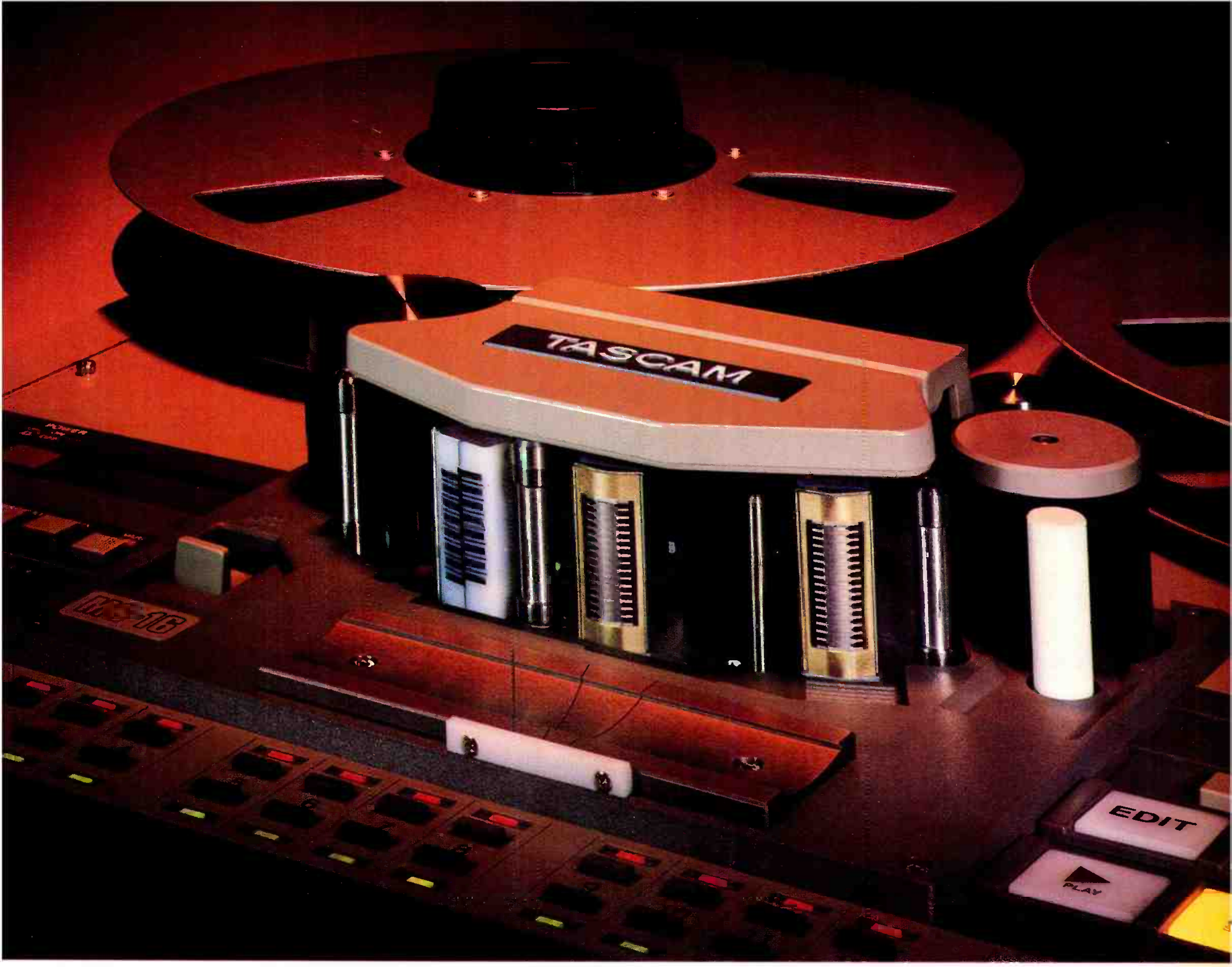
☐ **KAJEM RECORDING** (Gladwnye, Pensylvania) has upgraded its **Solid State Logic SL4000 E** console with a **Total Recall Computer**. The facility also purchased a **Yamaha REV-7** digital reverb. *Gladwyne, PA.*

*Midwest:*

☐ **STUDIO A** (Dearborn Heights, Michigan) has moved to a 4,000 square-foot facility, which is subdivided into a 700 square-foot control room, 800 square-foot studio, two isolation booths, and three live chambers. New equipment purchased for the move includes **Klark Teknik DN780** and **AMS RXM-16** digital reverbs, **Aphex Compellor, DeltaLab Compueffectron, Sony MCI JH-10C** two-track, **Neumann U-87** and **AKG 414** microphones, **Hafler P225** power amps, **Valley People Kepex II** noise gates and VCAs, and **Maxi-Q** equalizers, **UREI 813Bs, Yamaha NS10M** monitors, and **Roland JX80** synthesizer. In addition, **John Jaszcz** and **John Avedisian** have joined the studios engineering staff. *5619 N. Beech Daly, Dearborn Heights, MI 48127 (313) 561-7489.*

☐ **GNOME SOUND** (Detroit) has completed a major upgrade for audio and video production. The equipment list begins in the control room with a new 24-input **Soundcraft TS-24** mixing console, an **Otari MTR-90 II** 24-track master machine with EC-101 SMPTE synchronizer card, a **Roland SBX-80** Sync Box and **UREI Model 809** Time Aligned monitors. The next phase of the upgrade includes a host of MIDI gear, including a **Roland MKS-80** synthesizer module and **SVR-2000** MIDI-controlled digital reverb, **Akai** MIDI sampler, and an **Apple Macintosh** personal computer for MIDI sequencing. Related synthesizer purchases include a **Yamaha TX 216** — an expander rack for the exisiting DX-7 — a **PPG Waveterm 2.3** digital keyboard, and a **Linn 9000** drum synthesizer. *3034 Shenendoah, Royal Oak MI 48073 (313) 549-5286.*

☐ **STUDIO 95** (Cincinnati, Ohio) has opened a new production studio aimed at original music, jingle productions and radio/TV

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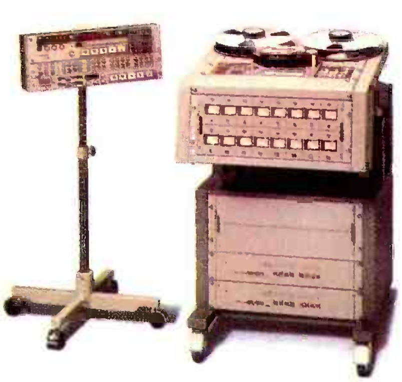

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**For additional information circle #121**

commercials in conjuction with the facility's parent company **W-LITE 95** Radio. The new LEDE certified control room will house an **Audiotronics 328** console, **UREI 813, JBL**, and **Fostex** monitors, an **Otari MX-5050B** eight-track with autolocator, **Sony MCI** four- and two-tracks, and a **Studer A80**. Outboard gear includes a **Yamaha RX-7** and **Lexicon PCM-60** digital delay and reverb, **Eventide Harmonizor, Aphex Aural Exciter, Valley People Dynamite** noise gate, **Orban** stereo synthesizer, and a host of microphoes from **Neumann, Sennheiser, Shure**, and **Electro-Voice**. In-house synthesizers include **a Yamaha DX-7** with **Model 816** slave modules, **RX-11** drum machine, and **QX-11** sequencer. *250 West Court Street, Cincinnati OH 45202 (513) 241-9500.*

□ **SEAGRAPE RECORDING** (Chicago) has opened as a 3,500 square foot 24-track facility designed by **Robert Jones**. Featured in the facility is a 56 by 50 by 24 **Neotek** console linked to **Sony MCI JH-24** and **-114** 24- and 16-track machines, respectively; a **Sony F-1** digital processor; an **Otari MX5050** half-track; and a **3M M56** 16-track. Various outboard comprises **Lexicon Model 200, Super Prime Time** and **PCM 60** digital reverbs; **Eventide Harmonizer 949; dbx Model 166** stereo compressor/limiter; and an **Echo Plate II**. Monitoring is provided by **JBL 4430** bi-radials, **4311s**, and **Yamaha NS10M** reference monitors. *5740 North Western Avenue, Chicago IL 60649 (312) 784-0773.*

*Southeast:*

□ **RON ROSE RECORDING STUDIOS** (Tampa, Florida) is a recently opened, three-studio audio-for-video complex. The facility has purchased a new **Yamaha DX-7** synthesizer, **Panasonic** and **NEC** half-inch VHS machines, and a **Sony 5850** three-quarter VCR. In addition, the company is starting a Compact Disc library consisting of 60,000 different songs and sound effects for its two complexes, the second of which is located in Detroit. *3409 Lemon Street, Tampa, FL (813) 873-7700.*

□ **APPLESON STUDIOS** (Miami) a radio/TV commercial production complex, has expanded its audio-for-video capabilities by adding a custom eight-track **Pacific Recorders ABX-34** stereo mixing console (featuring the latest **Dean Jensen** electronics for transformerless operation) to its Studio A, and seven new **Otari MX5050B II** two-tracks. An **Aphex Aural Exciter** type B and compellor/levellor system, a new patchbay, a new monitoring system — featuring **Hafler P-500 VMOSFET** power amplifiers —were also incorporated into the upgrade, along with a stereo monitor oscilloscope and phase compatibility meter. The last addition was the construction of a 2,000 foot office/studio to accommodate the facility's new Telemarketing program and studio/control room, which will feature a 16-channel re-built **ADM 2005** broadcast board. *Miami, FL (305) 625-4435.*

□ **CRESCENDO RECORDERS** (Atlanta) has added a keyboard programming room to its two-studio complex. Named Studio C, the new room will house ten **Yamaha DX-7**s, an **Emulator II**, a **Juno 106**, a **Drumulator** and **Oberheim DMX** drum machine, **Roland** guitar synthesizer, and a **Commodore 64** microcomputer for MIDI control and computer interface. **Gary Ham**, chief engineer, says the new studio may be used by outside artists for demos and record production, as well as for jingles, movie scores and record production. Also, added to Studio A was an **Otari MTR 90 MkII** 24-track, **Adams Smith 2600 Series** synchronizer, a **Publison Infernal Machine** digital reverb, **Fullmost** stereo enhancer/de-esser, and **Lexicon Model 200** digital reverb processor





For additional information circle #162

with six software programs. Monitors are custom **Acoustics Physics** cone speakers. The facility also says it is organizing a pro-audio equipment rental program to hire equipment ranging from 24-track machines to microphones. *125 Simpson Street NW, Atlanta, GA 30313.*

*South Central:*

☐ **DALLAS SOUND LABS** (Dallas) has redesigned its Studio C, offering post-production sound for video or film work. The new control room centers around a **Sony MCI 536** automated console. Tape machines include **Sony MCI JH-114, -110, -110-B** 24-, four-, and two-, and one-inch playback machines, respectively; **PCM 3324** 24-track digital, **Otari MX5050B** two-track, **Technics SV-1000** digital and **1520** analog two-tracks. A **BTX Softouch** synchronizing system interlocks the tape machine network. New outboard gear comprises the following reverb units: **AMS, Yamaha R1000,** and **BAE** plate. An **Eventide 949 Harmonizer** and **Phasor** are also utilized for effects purposes. An **Allison Kepex** and **Gain Brain, dbx Model 165** noise gate, **Valley People 430, Aphex Compellor, Biamp** one-third octave equalizer, **Orban 622-B** parametric equalizer and de-esser round out the outboard array. **Dolby** and **dbx supply** noise reduction, and control room monitoring is supplied by **Yamaha NS-10Ms**. In-house digital synthesizers include the **Kurweil 250, Yamaha DX-7, Korg Poly 800,** and **Simmons** digital drums. *6305 N O'Conner Road, Suite 119, Irving TX 75039 (214) 869-1122.*

*Southern California:*

☐ **HANNA BARBERA** (Hollywood) has purchased a customized 32-input **Westar** console and **Quad-Eight Electronics** outboard processing equipment from the **Mitsubishi Pro-Audio Group** for its in-house dubbing facilities. According to the facility's sound director **Alvy Dorman**, "This purchase will bring the studio to a comparable, sophisticated level with its

RON ROSE — opens Tampa facility

SWINGSTREET — new 24-track studio

HANNA BARBERA — custom Westar

computer-oriented animation department." The two-man desk (music/dialog and effects) is divided into two 16-input mix positions with monitor and motion controls located in the center, and which features 33 60-segment LED bar meters. The peripheral outboard processors include **CL-22DS** compressor/limiter/de-essers, two **EQ-712** graphic equalizers, two **VFX-200** variable effects filters, and two three-channel pan pots. Also provided are two A/B transfer keys and ten cartridge start/stop remote controls. The decision to purchase this modular console coincided with the facility's decision to open its dubbing suite to independent producers during its "off season." *3400 Cahuenga, Hollywood, CA 90068 (213) 851-5000.*

☐ **VOICE OVER L.A.** (Hollywood) is a new two-studio eight-track facility geared for radio, televison, and film productions. Studio A, measuring 15 by 14 feet with a 15 by 12 control room, centers around a 24 by 16 by 2 **Soundcraft 1600 Producer Series** console. Studio B, measuring 13 by 12 with a 12 by 8 control room, centers around a 16 by 4 by 2 **Biamp 1642** console. Tape decks within the complex include an **Otari 5050 MkIII** eight- and four-track, four two-tracks, and two **Studer Revox PR 99s**. Four **Hafler P225s** and four **Symetrix A-220s** drive two pairs of **JBL 4311s**, three pairs of **Aurotones**, and two **Sony RPM TV** speakers. Outboard gear comprises a **Lexicon PCM 60** digital reverb, **Korg SDE 3000** digital delay, **dbx Model 166** stereo limiters, **LT Sound** vocal eliminator/reverb control center, **Technics SL-P50** Compact Disc player, and an **Audio Kinetics Q.Lock 3.10** synchronizing system. Microphones from **Sennheiser, Schoeps, Shure** and **AKG** round off the list. *1717 North Vine Street, Hollywood CA (213) 463-VOLA.*

☐ **WESTLAKE STUDIOS** (Hollywood) has opened Studio D for film scoring, album projects, and audio-for-video production. From a converted warehouse that formerly housed audio equipment for the dealership portion of the facility, Studio D's 500 square foot control room boasts a new 56-input **Harrison MR2** console equipped with **Audio Kinetics Mastermix** automation, currently linked to a **3M 79** 24-track machine. Supplementing tape decks and outboard gear will be drawn from other Westlake studios. The studio portion of the room measures 1,000 square feet. In addition, there are two isolation booths: one of which is specifically for drum and percussion work, while the other serves as an "echo chamber" with up to one second of decay attainable; an electric drape system, consisting of heavy velour drapes, allows acoustic tailoring of the mid to high end of that room. Production offices, a tape libaray, a machine room, and a producer's projection room are also available to clients when they use Studio D. *7265 Santa Monica Boulevard, Los Angeles CA 90046 (213) 851-9800.*

*Northern California:*

☐ **SWINGSTREET STUDIOS** (Sacramento) is a new 24-track facility designed by **Jack Edwards** and tuned by **George Augspurger**. The control room features a 28 by 24 **Quad-Eight Pacifica** console, a **Sony MCI JH-16** 24-track, and two **JH-110B** two-tracks. Outboard gear includes a **Lexicon 224** digital reverb and **Super Prime Time**, two **UREI 176LN** limiters, two **Teletronix LA2A** limiters, **Allison Gain Brains, Aphex Aural Exciter, Eventide H910 Harmonizer, AKG BX20** reverb and various microphones, and **Pultec PEQ-1S** equalizers. Monitors are supplied by **JBL, UREI,** and **Auratone**. *620 Bercut Drive, Sacramento, CA 95814 (916)446-3088.*





For additional information circle #164

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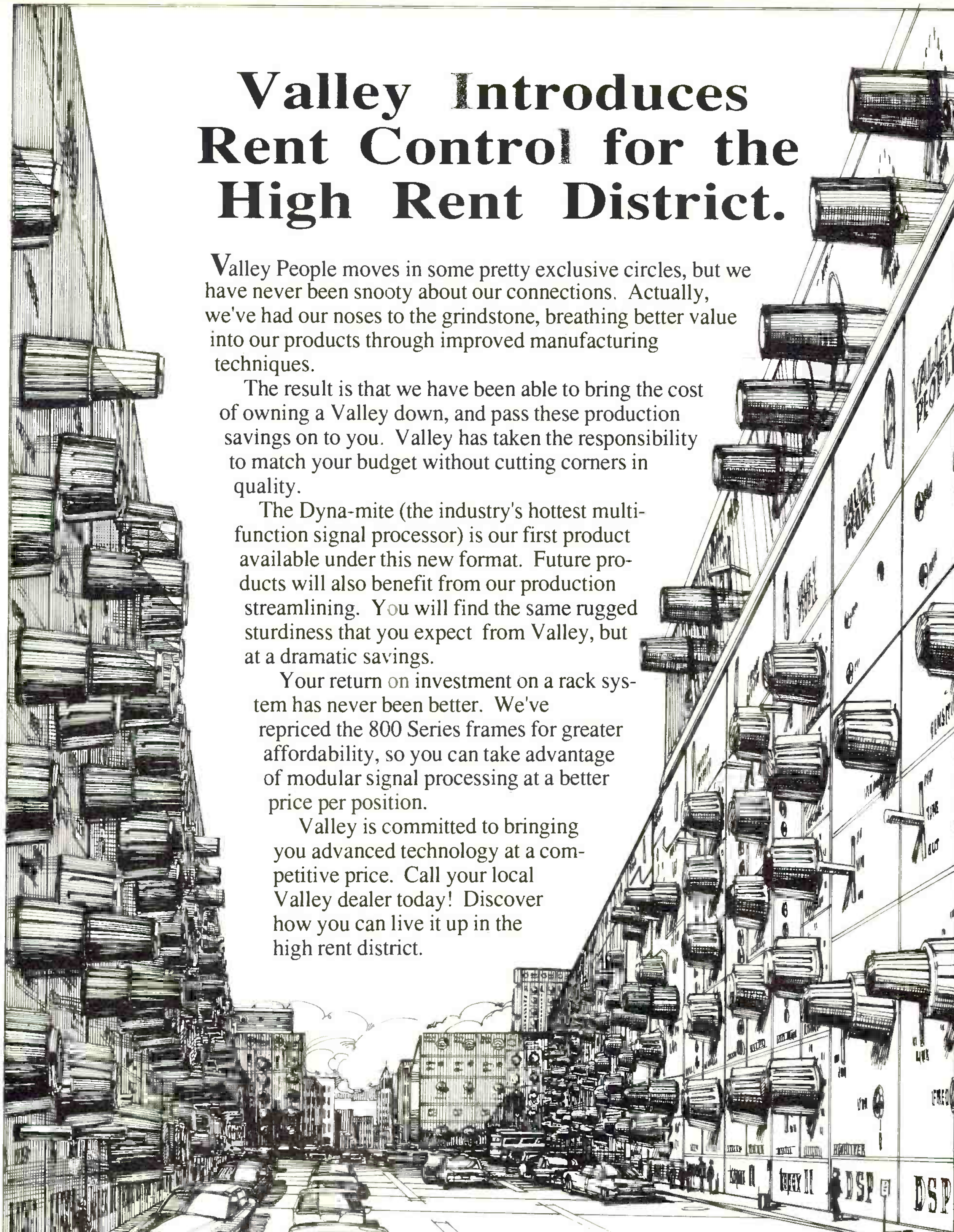


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**For additional information circle #165**

# STUDIO UPDATE

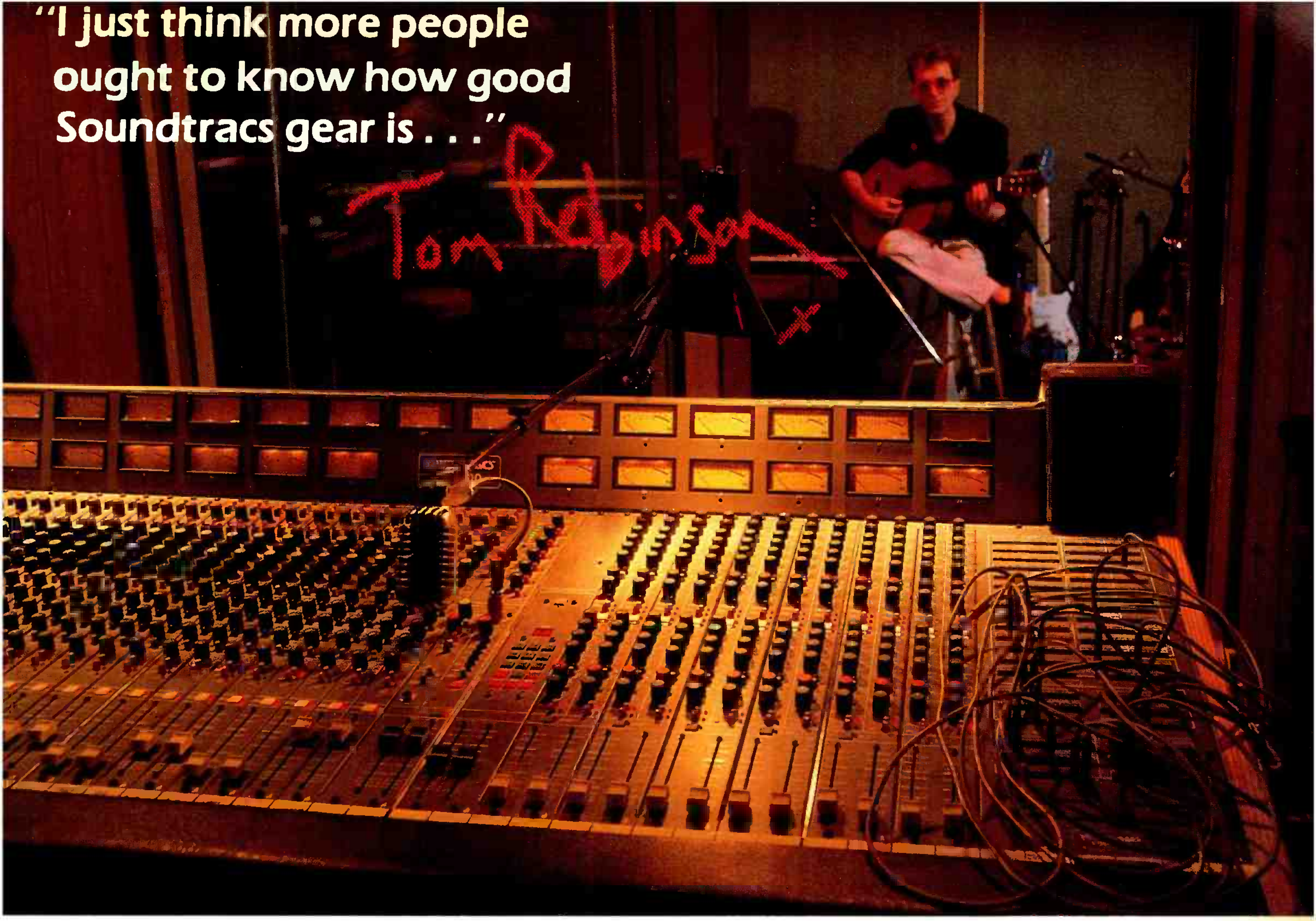
FACILITIES
EQUIPMENT
PEOPLE

□ **LIVE OAK STUDIO** (Berkeley) has added a **Sound Workshop Disc Mix II** IBM controlled disk-based mixing automation system, and digital sampling software upgrades to its existing **Kurzweil 250** synthesizer for use by the **Macintosh** computer for programming and storage. *Suite #364, 1442A Walnut Street, Berkeley, CA 94709.*

## STUDIO PROFILE: Full Sail Opens In-House 24-track Facility

□ **FULL SAIL RECORDERS** (Altamonte Springs, Florida) has opened a new 24-track facility to serve as both a commercial music studio and on-site student workshop, as a joint venture between **Full Sail Center for the Recording Arts** and **Parc/CBS** — an independent record label based in Orlando, FL. Designed by acoustician **John Storyk**, the 40-by-30 foot studio and 25-by-25 foot control room (which boasts a frequency response of ±2.5 dB from 40 Hz to 6.3 kHz, prior to room EQ) houses a 56-input **Solid State Logic SL 6000 E** console with **Primary Studio Computer** and **Total Recall** automation with SMPTE timecode capabilities. Linked to the SSL are a **Studer A800** 24-track, a pair of **A80** two-tracks, and a **A80** four-track. The studio playback system comprises a pair each of **Fostex LS-3**s and **RN 780**s; control room reference speakers consist of **Fostex LS-2B**s, **Yamaha NS-10**s and **Auratone 5C**s. A full range of outboard gear includes **Lexicon 224XL, Model 200,** and **AMS RMX-16** digital reverbs; digital delay lines supplied by an **AMS DMX15-80S**, two **Lexicon PCM 42**s and a **Prime Time,** an **Eventide 969 Harmonizer,** a **DeltaLab CompuEffectron 1700**, and two Lexicon PCM 41s; noise gates from **Drawmer** and **dbx;** an **Aphex Compellor;** two **dbx 900** racks; noise reduction systems from **Dolby** and **dbx;** and **UREI LA4A** compressor/limiters. A host of microphones from **Neumann** and **Senheisser** round out the studio's equipment list. **John Phelps** president and director of the Full Sail Center for the Recording Arts says "the facility now has the most technically advanced console in Florida," making the school the only "Florida-based [institution] to offer hands-on training on such technically advanced equipment as an SSL." This advantage, he feels, will give students a competitive edge in the professional audio job market. Referring to other facets of the complex, a 24-track mobile — called **The Dream Machine** — is available for location recording, as well as a room called **Digital:One** — a realtime cassette duplication facility featuring two **dbx Model 700** digital two-track processors linked to **Sony 5800** VCRs (analog half- or quarter-inch masters can also be transferred from an Otari MTR-12 two-track). *660 Douglas Avenue, Altamonte Springs FL 32714 (305) 788-2450.*

**FULL SAIL — new SSL and Studer studio**




"I did a lot of research before buying a mixing console for Turbot studios. Ideally I'd have liked an SSL or a Harrison I suppose, but frankly, they were out of my price range.

Among the producers & engineers I asked, however, the general opinion was that — if it lived up to its claims — one of the new Soundtracs desks would be excellent value for the money I had.

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Like Pete Townshend I paid the normal retail price for my desk and didn't get any backhanders for this ad. I just think more people ought to know how good Soundtracs gear is . . ."

**TOM ROBINSON**

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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) 640 6447
In Canada: **Omnimedia Corporation Ltd.** 9653 Côte de Liesse, Dorval, Quebec H9P 1A3. (514) 636 9971



For additional information circle #170

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MARK CUSTOM RECORDING SERVICE
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(212) 541-5022 DM

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Charlotte, NC 28209
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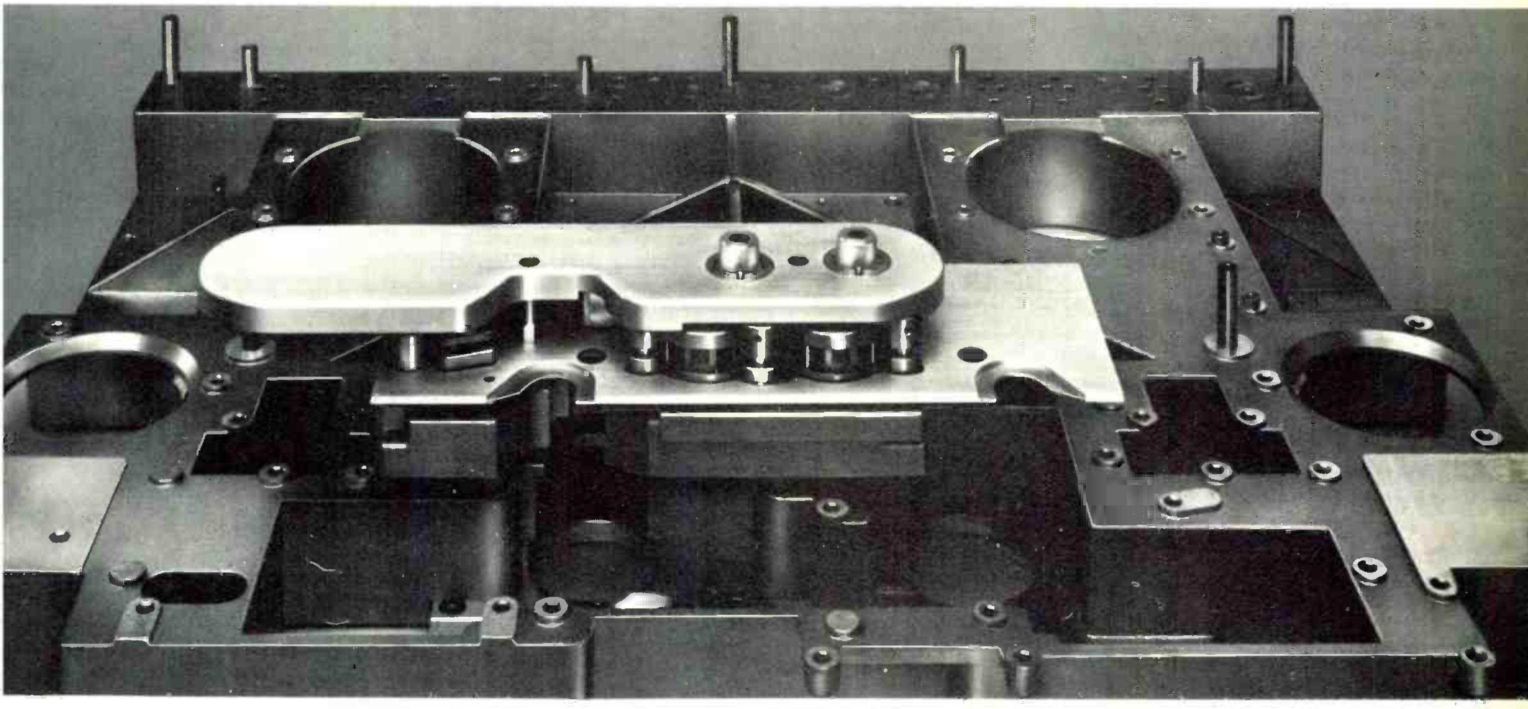
Studer A80MR MKII and A80QC MKII



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Stability and precision are essential at both ends of your cassette duplication chain: mastering and quality control. Your cassette product is only as good as your master, and your quality control is only as good as the machine performing it. That's why leading duplication facilities – including Capitol, CBS, MCA, and RCA – rely on mastering and QC recorders from Studer.

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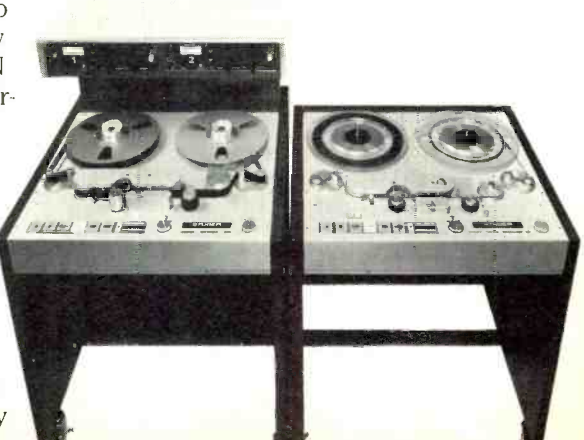
*Dolby HX Pro is a trademark of Dolby Laboratories, Inc.

STUDER REVOX



For additional information circle #173

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Glendale, CA 91201
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Van Nuys, CA 91406
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# RECORDING AND PRODUCTION APPLICATIONS OF DIGITAL SAMPLING DEVICES

## Triggering Sampled Sounds from MIDI, Control-Voltage and Signals to Replace Drum, Vocal and Similar Tracks

*by Denis Degher*

**SAMPLE,** *n., v.,* —*n.* **1.** a small part of anything or one of a number, intended to show the quality, style, or nature of the whole; specimen; —*n.* **2.** serving as a specimen; —*v.t.* **3.** to take a sample or samples of; test or judge by sample.
**SAMPLING,** —*n.*, the act or process of selecting a sample for testing, analyzing.
**TRIGGER,** *n., v.,* —*n.* **1.** anything, as an act, event, etc., that serves as a stimulus and initiates or precipitates a reaction or series of reactions; —*v.t.* **2.** to initiate or precipitate a chain of events; —*n.* **3.** a device, as a lever, the pulling or pressing of which releases a detent or spring; —*n.* **4.** a small projecting tongue in a firearm, which when pressed by the finger, actuates the mechanism that discharges the weapon.

In the recording environment, sampling is the technique of digitally storing a sound sample, and then triggering or manually or electronically keying replay of the sample to achieve the desired result or effect. Today's marketplace is beginning to offer a vast variety of sampling keyboard instruments, including the New England Digital Synclavier, Fairlight CMI, PPG Waveterm, E-mu Systems Emulator II, Kurzweil 250, Ensoniq Mirage, Sequential Prophet 200, and a new generation of sampling black boxes. Because of the growing importance of sampling techniques, this article will concentrate on the use and application of non-keyboard-oriented sampling tools; an article by Quint Randle published in the October issue of *R-e/p* provided a handy overview of the use of sampling keyboards in the studio.

Today's sampling devices are based on the mushrooming advances and cost savings being made in computer technology and, like an inverted pyramid of ever-expanding knowledge, new devices and capabilities will begin to appear at an increasing rate. Like the ever proliferating hardware, software advances will enable us to store longer samples. And, with the falling cost of microprocessors and random-access memory, the ability of such devices to store and retrieve data promises almost infinite storage time in the not too distant future. While current sampling devices utilize RAM- or EPROM-based storage, future systems will allow for multitrack information to be stored on hard disk for extended editing and manipulation.

The present generation of sampling devices include upscale units like the **Advanced Music System** DMX 1580S digital delay with pitch change and sampling capabilities; the **Eventide** SP2016 Generation II Effects Processor/Reverb with a loop-edit sampling mode; and the **MDB Synthesizers** Window Recorder with overdub function for sound-on-sound samples. Other devices presently available include the **Analog & Digital Systems** DeltaLab CompuEffectron CE1700 (see accompanying sidebar for a simple modification to provide external triggering of sampled sounds); **Publison** Infernal Machine; **Korg** SDD-2000 sampling digital delay; and the **Akai** S612 Midi Digital Sampler with MD280 floppy-disk. Another device that has been shown in the prototype stage at several recent trade shows is the **Oberheim** DMS Digital Midi Sampler/EPROM Programmer.

While all these units have the common ability to store sound samples, that's where their similarities end since each manufacturer offers varying features and aims for a different niche in market share.

The 16-bit AMS DMX 1580S was possibly the first commercially available device to offer sound sampling. The 1580S was first introduced in late 1981 as a digital delay equipped with a Hold function, and the ability of editing the sample to obtain the desired sound. In 1983, the company unveiled an audio triggering update, and sound sampling with external triggering was born.

The two-channel unit can simultaneously provide level-conscious triggering in channel A, and pitch changing in channel B. Channel A's four-PCB card slots enable either 1.6- or the 3.2-second memory cards to be installed with optional memory capacity of up to 12.8 seconds. (Custom-modified units may be ordered with sample capacity of up to 23 seconds.)

October saw the release of a new update from AMS that enables sampling and triggering of *two* independent sounds.

When the 1580S first appeared, a typical use was the replacement of snare or bass-drum sounds. But, with increased sampling times becoming available, longer instrumental phrases and vocal hooks can now be stored and triggered, a development that heralded a new era in recording.

Another interesting function of the 1580S is the use of channel B's pitch-change circuitry; after the sample has been loaded, the pitch changer can be used like a VSO to enhance the sampled sound. The modular unit is software updatable, and offers nine, non-volatile user-memories, plus optional keyboard Interface/Chorus Controller; a MIDI interface card is reported to be available soon.

The 16-bit MDB Synthesizers Window Recorder sampler, manufactured in Switzerland and available in the U.S. from Europa Technology, comes standard with six-second memory, and is available with an optional 12-second sample capacity. Sampled sounds can be off-loaded onto floppy diskettes or into a computer via a 37-pin communications buss. The unit features a quoted 96 dB of dynamic range, a frequency response of ±1 dB,

. . . *continued overleaf* —

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20 Hz to 20 kHz, and a 44.1 kHz sampling frequency.

Once the sound has been sampled into the Window Recorder, its pitch can be raised or lowered by one octave, and controlled harmonically or dynamically by any keyboard, computer, or drum machine via a control-voltage input. Another interesting capability of the Window Recorder is an overdub function that enables the mixing of the original sample with another incoming sound, to create a ping-ponged or sound-on-sound effect. The unit was designed strictly as a sampling device, and the simple control-panel layout makes sampling a very straight forward process.

The Eventide SP2016 Generation II Effects Processor/Reverb is a highly sophisticated unit that now offers sampling in the loop-edit mode. [A complete system description was included in Bob Hodas' review of the SP2016 published in the April issue of *R-e/p — Editor.*] Although the unit has been on the market for a couple of years now, the software-based system has continually evolved, making it one of the most versatile and advanced devices around. The single-channel sampling program provides 1.63 seconds of sample time at a 16 kHz bandwidth, or 3.26 seconds at an 8 kHz bandwidth. The sound segment can be edited from either the front or the back, and the sample triggered from the front panel, remotely, or from an audio input trigger.

The new Oberheim Digital Midi Sampler/EPROM Programmer should be available by the end of the year, and samples sounds up to 5.33 seconds at 12 kHz, or just over two seconds at a 32 kHz sampling rate. The unit uses 8-bit companding, 64 kbytes of RAM, and is described as being compatible with any MIDI-equipped keyboard; sampled sounds are playable over the full MIDI range with velocity control. The device features digital editing of the memory contents, and the ability to "record" sounds onto EPROM (Erasable Programmable Read-Only Memory) chips, and then use them in other compatible devices. The DMS also reads most EPROMs available from other manufacturers, and has the ability to manipulate the sounds and mix the result with another sample, to create completely new sound textures. The DMS promises something for both the musician and the studio engineer.

The Korg SDD-2000 is a sampling digital delay line compatible with a MIDI-equipped keyboard, drum machine, or sequencer. In the sampling mode, recorded sounds can be actuated via MIDI control, or audio triggering. The SDD-2000 can increase the pitch of a sampled sound up nearly three octaves, and also features a MIDI-controlled sequencer mode. The unit has a quoted frequency response of 30 Hz to 18 kHz at a sampling capacity of 1.092 seconds, or 30 Hz to 4.5 kHz at 4.368 seconds.

The Akai S612 Midi Digital Sampler

*. . . continued on page 108 —*

## HINTS AND TIPS FOR SAMPLING, EDITING AND TRIGGERING DIGITAL DELAY UNITS AND SAMPLERS

*by Joe Chiccarelli*

In terms of procedures, the actual recording of sound samples is one of the simplest steps in the entire sampling process. Most of the sampling units I've encountered take a simple line-level input (anywhere from -20 to +4 dB), which means that if you're using a microphone or other mike-level signal as your input, you will have to use a pre-amp to bring the signal up to the appropriate level. From there, it's just a simple matter of hitting a "Hold" or "Lock In" switch (different units are labeled differently) to actually get the sound into the box. On some units, you hit the switch *before* the sound occurs, while on other units, you hit the switch *after* the sound occurs.

**— Engineer/producer Joe Chiccarelli —**

The most important thing at this stage is to achieve the best possible audio quality in your sample. You should take all the care that you would normally take in recording a sound live to tape, in terms of microphone technique and eliminating unwanted sounds such as fret noise, air-conditioner hum, chair squeaks, etc.

Since most samplers have limited headroom and dynamic range, you want to be careful about signals that have very sharp transients — such as drums and percussion instruments. You might encode a sound with the sampler set for maximum headroom and maximum transients; but then, when you play your sample back, you find you have very little VU level and a lot of digital noise from the sampling unit itself. Sometimes a little bit of compression can help you around all of these difficulties. Another approach is to first record the sound on analog tape (preferably 30 ips, half-inch), which helps smooth out the transients, and lets you pack the sound into the sampling unit at a slightly higher level.

A further advantage of first putting sounds on tape is that you always have them on hand to resample later for another song, another project or whatever. In some cases though, you may prefer the immediacy of getting a particular sound right into the sampler. Both approaches are equally valid. It's just a matter of your preference, and the particular circumstances of what you're doing.

### Editing Sound Samples

Once your sound is actually in the sampling unit, the next step is to edit it. What you're doing here is changing the amount of digital storage time in the sampler. Say you're using a unit with a sampling capability of five seconds, and that the sound being sampled is a snare drum. The average snare drum sound is only going to be a few hundred milliseconds in length. So, depending on at what point you hit the "Record" control while sampling the sound, you're probably going to have to take time off the front of the loop and off the back. physically, this involves using a "Nudge" button or — in the case of samplers that are also digital delays — the unit's delay-time controls.

The main reason for editing the front end of your sample is to ensure that the attack of the sound begins at the *precise* instant at which it is triggered — either by an audio signal from tape, a musical keyboard (in the case of MIDI-equipped samplers) or the "Play" button on the sampler itself. You edit time off the back end of the sample to get rid of any extraneous noise, such as a tape machine shutting off, drum sticks clicking, etc.

But, apart from these basic functions, editing provides you with a means of creatively manipulating your basic sampled sounds. If you have a keyboard sound with a slow attack time, for example, you can chop off some of that attack time when you edit the sample, thereby changing the basic shape of the sound itself. Similarly, you can alter the envelope of a sound by lopping off some of the decay as well.

### Manipulating Pitch

Some sampling units also let you alter the pitch of your sample, providing even more creative control. Some samplers are actually pitch shifters, and there are also many MIDI-based samplers that let you use a keyboard instrument to manipulate pitch. These pitch-shifting capabilities are especially handy while working with vocal tracks. If you have one or two words in a song where the pitch is a little off, you can easily correct them. You

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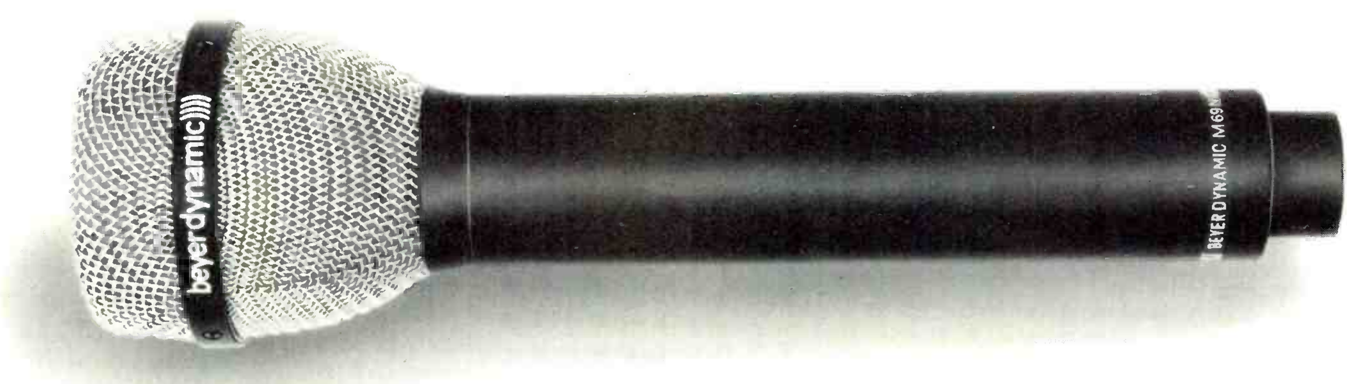


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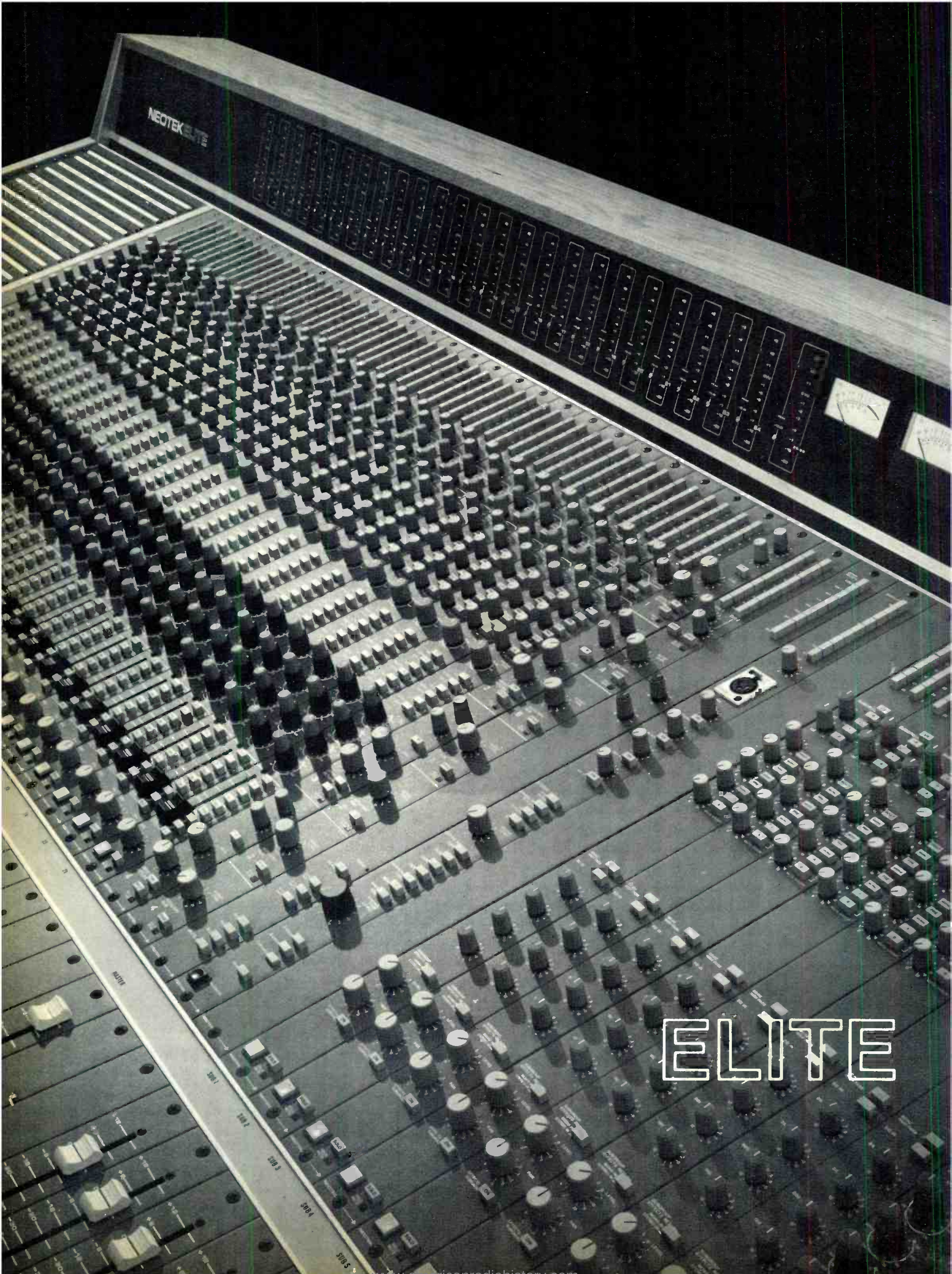
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# SAMPLING HINTS AND TIPS — continued . . .

can even sample entire phrases and alter the pitch slightly to create some wonderful doubling effects.

Conceivably, you can even create harmonies, but here you're really limited to one or two notes because of the time factor involved. As you compress or expand a sampled signal to alter the pitch, you are also compressing or expanding the duration of the sample. In other words, if you have a 500-millisecond sample, and you shift it up an octave, it will actually turn into a 250-millisecond sound. And, of course, the opposite is true: if you lower the pitch, the duration of the sound becomes longer.

The other thing to remember about shifting the pitch of samples is this: if you sample a C and then move it in pitch down to a B, it's never *really* going to be a B, because all the overtones associated with a B are different from those associated with a C. So, when you move that C down in pitch, the partials are just going to move down in a strictly parallel manner, as opposed to creating a *new* overtone series (that of a B), as would happen if you actually played a new note. The unrealistic sounds you get as a result of these kinds of pitch shifts can be an asset or a liability, depending on what kind of effect you are trying to create.

### Triggering the Sampled Sound

One of the most common uses of sampling is to replace a questionable sound that might already be on tape; maybe you have the greatest take in the world, but the snare drum went out of tune toward the end of the song. Or maybe the drummer hit the snare just a little off center in one very crucial spot, and it's driving you crazy. Thanks to sampling, you can lift a good note or phrase from your almost-perfect take, and use it to replace any bad notes that may exist.

In each of these cases, replacing the bad notes involves using the existing track to trigger the sampler, which has been loaded with the new sound that you want to use. Some sampling devices provide a separate trigger input, and on others you use the same input for triggering that you used to store the sound in the first place. Most samplers will accept anything from -10 to +4 dB (0.3 to 1.6V) as a trigger. On various projects that I've done, I've used everything from snare-drum tracks to vocal tracks to trigger a sampling unit. All you have to do is adjust the input level in such a manner as to avoid "double triggers," or other problems stemming from the sampler's over-sensitivity to the trigger input.

Apart from this, microprocessor delay is another factor that makes triggering the trickiest aspect of using samplers. Basically, the delay results from the time it takes for the microprocessor within the sampling unit to figure out exactly what it is you're doing with the unit; there can be anything from two to 14 milliseconds (in the case of some of the older units) between the trigger signal arriving at the sampler, sound being produced.

Fortunately, there are several ways of correcting these time delays. If you're in a mixdown situation, and you're using a tape machine that has separate sync and repro outputs, the time delay problem is easy to correct. You simply take the sync-head output and use that as trigger for your sampler. Now, there is a time delay between the sync and repro outputs of most tape machines. (On the Studer A-800, for example, the sync-head output is around 51 or 52 milliseconds ahead of the repro, but the figure will differ on other machines, depending on the physical distance between the sync and repro heads and the tape speed.) The simplest thing to do is to place a delay unit in the line between the sync (trigger) output of the tape machine and the trigger input of the sampler. Depending on your sampled sound, all the other sounds on the track, and the overall feel of the track itself, you will have to audibly tweak the delay unit that's on your trigger in order to bring the sampled sound in sync with all the other sounds on the track.

If you want to trigger a sample while you're overdubbing (i.e., when you're in the sync mode rather than the repro mode), there is another, slightly more cumbersome, approach you can take in order to correct time-delay problems in triggering the sampler. Say you're using a snare sound as your trigger. What you do is this: flip over your multitrack tape and record the snare onto an open track, passing the signal through a delay unit set at a ballpark delay time of 20 to 30 milliseconds. When you flip the tape back over and play it back normally, you will have the delay on this newly recorded snare track occurring *before* the snare itself. Using this pre-delayed snare track as a trigger, you can overcome the microprocessor delay produced by the sampling unit.

But suppose there's too much delay on the trigger. In order to correct that, you have to take a second delay unit and place it between your trigger track and the delay unit's trigger input. Using this delay, you can — once again — fine-tune your trigger signal, placing the sampled sound ahead of the beat, behind the beat or right on the beat, depending on what you want.

In 48-track situations, a third approach is to copy the trigger track from the master machine to the slave, and program a timecode offset into the synchronizer that will compensate for microprocessor delay within the sampling unit.

On the whole, then, microprocessor-delay problems involved in triggering samples are fairly easy to overcome. While the solutions to these problems may be a little combersome, they're a small price to pay for the amount of creative freedom that digital sampling offers. □□□

— continued from page 105 . . .

## CREATIVE APPLICATIONS OF SAMPLING DEVICES

and optional MD280 disk-drive storage unit does precisely what its name suggests. Although the device does not come with a keyboard, it is intended for use with MIDI-equipped keyboards, and thus has the ability to track notes with velocity. The device uses 12-bit sampling, and can function as either a mono or polyphonic (six-voice) sampler. It also has the ability for overdubbing sounds on top of one another to create stacking or layered sounds — this latter function is made easier by the provision of an auto-trigger control in the sampling mode.

Just as the future always merges with the present in the space/time continuum, our current studio hardware forms a basis for tomorrow's technology. Just when state-of-the-art seems to have been defined, it becomes old hat like the term itself, as recently developed technology again moves to the forefront. With sampling currently in vogue throughout the music business, it is only a matter of time before the creators of ***Future-Tech*** will again astonish us with the next new processor. Until then, the world of sampling will offer manifold ways to enhance or possibly obfuscate the creative process, so that it too becomes tried and true proven technology. Happy sampling. ■■■

## PRACTICAL APPLICATIONS OF SAMPLING IN THE STUDIO

My first experience with sampling in the control room was a manifold experiment in the joys of a new technology that obviously will have a broad impact on the future of the recording industry. As sampling times lengthen, more and more ways to use this new technology will unfold and, akin to the invention of new paint pigments during the Renaissance, will have far a reaching affect not only on how producers and engineers perceive record making, but the ways in which artists themselves approach their contribution to the creative process.

With at least one unit presently offering 23-second sampling times, many artists, producers and engineers will consider it common practice for vocalists to sing just one chorus, and merely use a sample to create the rest of the song. No one yet knows what impact this capability eventually will have on the creative integrity of artists but, as the technology further entwines itself into the creative fabric of modern studio life, profound changes indeed will occur.

As with any new high-tech device, removing them from the box can be

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## CREATIVE APPLICATIONS OF SAMPLING DEVICES

an exciting and sometimes disorienting experience. But to my surprise, the **AMS DMX 1580S**, the **MDB Window Recorder**, and the **Eventide SP2016 Generation II Effects Processor** — the three sampling units that I selected as being representative of currently available devices — were all fairly straight forward and user friendly to operate, although each unit had its own particular quirks and personality.

I had the opportunity to use all three units back to back during the recording of Patrick Gammon's new album, tentatively titled *Emergence*, produced by Bob Monaco. The high "techno-electronic" album afforded me the perfect chance to maximize the technical, as well as creative, use of these new toys.

Replacing the electronic snare drums with the multiple-use DMX 1580S digital delay line and pitch changer was my first sampling assignment. After a suitable snare sound sample was found, I loaded it into channel A and began the editing process.

Electronic editing is similar to tape editing, in that I had to edit down the head and tail of the 3.2-second sample window to approximately 200 milliseconds, the actual length of the snare sample we were interested in. This proved to be difficult, because there are no visual displays on the DMX front panel to aid in the editing process. Instead, one must use the keypad or "nudge" buttons to electronically reduce the window by ear, without clipping either the attack or decay of the snare sample. (However, if the sample *was* clipped it is not necessary to resample the sound because of the redundant memory on the unit.)

The unit's pitch-change function was very useful for tuning the snare pitch to the track, simulating the work of a real drummer.

Because of a five- to seven-millisecond microprocessor access-time delay — the time taken for the AMS to produce a sound in response to a triggering input — a special trigger track had to be created. The required "pre-delay" on the trigger was accomplished by flipping over the two-inch tape reels, and transferring a 30-millisecond delayed snare from the track being used to trigger the AMS. By flipping the tape over and recording backwards on an empty track, we were actually putting the trigger snare approximately 25 milliseconds *ahead* of the actual snare signal recorded on the multitrack. After this pass was completed, we flipped the multitrack tape back to normal.

Since the trigger was now *ahead* of the track, we ran the trigger track back through the DDL and into the AMS 1580S, and, by using a Lexicon Super Prime Time in delay mode, we rhythmically tuned the sampled snare to the track and recorded it on an empty track.

While this procedure can be time consuming, it does provide the option

**Application Note:**

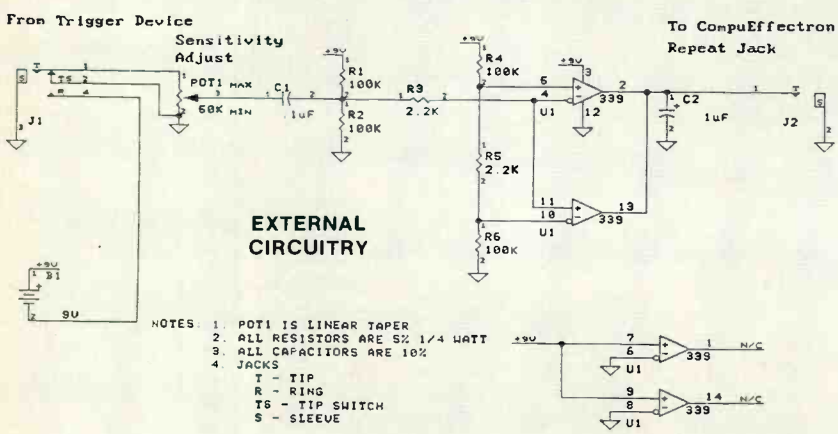
### EXTERNAL TRIGGERING MODIFICATION FOR THE ANALOG & DIGITAL SYSTEMS DELTALAB COMPUEFFECTRON CE1700

The ADS/DeltaLab CompuEffectron Model CE1700 can be used as a digital sampler capable of recording, editing, and playing up to 1.5 seconds of sound. The recorded segment can be played back remotely via a footswitch connected to the Repeat jack on the CE1700's rear panel.* However, it is often desirable to trigger the Play function from an audio-signal trigger rather than a footswitch. This application note, courtesy of ADS/DeltaLab, describes a simple circuit which accomplishes exactly that function.

The circuit schematic shown in Figure 1 needs no special precautions to be taken in construction. The Sensitivity Adjust should be a linear-taper pot, and should be brought out as an external control. The input jack is a stereo (three-conductor) type, and connects the battery when a mono (two-conductor) plug is inserted. (Therefore, do not leave a cable plugged into the input when the device is not in use.) A parts list with Radio Shack catalog numbers is included at the end of this application note. The entire circuit can be built in a simple enclosure for under $10.

Operation of the trigger interface is simple. First, record the segment to be played. When the segment has been edited as desired, recall preset #900 (press Recall, 9, 0, 0, Enter). Next, connect the input and output jacks to the proper places, and turn the sensitivity control to minimum. With an audio trigger signal present, slowly increase the sensitivity until the recorded segment begins to play in time with the audio-trigger signal; drum machines make very good trigger sources, as do piezoelectric trigger pads or drum pads.

□□□




| Designator | Description | Radio Shack Part# |
|---|---|---|
| U1 | LM339 Quad Comparator | 276-1712 |
| C1 | 0.1 uF Capacitor | 272-1069 |
| C2 | 1 uF Tantalum Capacitor | 272-1434 |
| R1, R2, R4, R6 | 100K Resistor | 271-1347 |
| R3, R5 | 2.2K Resistor | 271-1325 |
| POT1 | 50K Potentiometer | 271-1716 |
| J1 | ¼-inch jack, three-conductor | 274-261 |
| J2 | ¼-inch jack, two-conductor | 274-258 |
| B1 | 9V Battery | 23-464 |

**Note:** The information contained herein is believed to be reliable, but no responsibility is assumed for inaccuracies. Circuit diagrams are included to illustrate typical circuit applications, and do not necessarily contain complete construction information. Furthermore, this information does not convey any license under the patent rights of DeltaLab Research, Inc. or others. 

*Normally, a footswitch connected to this jack has the Repeat function, i.e., it behaves in the same way as the Repeat keypad. By recalling preset #900, a switch connected to this jack behaves the same as the Record keypad when neither the Repeat nor Record LED is lit, and behaves the same as the Play keypad when either is lit.

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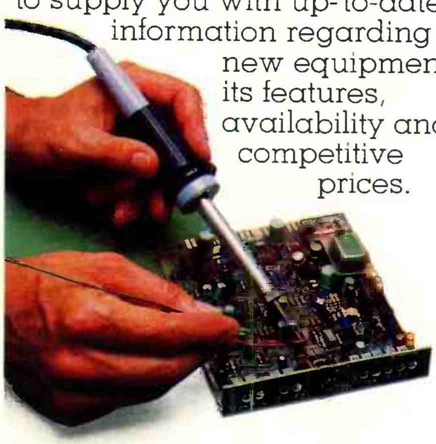

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Photography at Mad Hatter Studio, Los Angeles, by Duncan Aldrich

**Author Denis Degher adjusting the MDB Window Recorder.**

of placing the snare beat *precisely* where you want it, be it right on top of the beat or, in the case of the song "Never Be Happy," we placed it two milliseconds behind, giving it the slightly "layed-back" feel that a real drummer might play on a mid-tempo tune. (By doing it this way, we gave the song a human feel and a real snare sound, taking it strictly from the ranks of pure drum-machine feel.)

On several other songs, we placed the sampled snare directly on top of the beat to maintain the pure "techno-feel." Once the triggering procedure has been established, running trigger tracks becomes a very simple and matter-of-fact procedure, making the great sounding AMS 1580S fun to use.

Since the unit I had for evaluation was fitted with a 3.2-second memory, I was restricted to using it for drum samples. For sampling longer sounds, I employed the MDB Synthesizers Window Recorder, which was fitted with a six-second memory; a 12-second version is available as an option. This sampling-only device does its one job exceptionally well, and the unit's minimum front-panel controls make it easy to use. (Which was perhaps fortunate, because we were not provided with a user's manual, although my assistant Duncan Aldrich and I still managed to figure it out very quickly.)

The Window Recorder has an exceptionally fast processor access time of 50 microseconds, making it unnecessary to flip the tape and print advanced trigger tracks. It also features visual-assist editing, denoting the sound sample and edit parameters in green and red LEDs, thereby simplifying the editing procedure immensely.

In the auto-trigger mode, the front end of the sample is edited automatically, because the unit begins sampling on an audio trigger, which makes editing the tail the only real task to perform.

On the remake of the song "Tell Me Something Good" (originally produced by Bob Monaco), we recorded the vocal phrase, "Tell Me, Tell Me," through a Lexicon Super Prime Time digital effects unit with flanging onto a two-track. The vocal line was then processed through an ARP 2600 by synthesist/arranger Bruce Lowe, creating an enveloped, three-voice effect. This sound was then sampled into the Window Recorder, and manually inserted back onto the multitrack tape in the appropriate places.

Another interesting feature of the Window Recorder is its instant reverse function, which makes the production of backwards sounds a very simple process. Other interesting features include MIDI control and CV (Control Voltage) ports on the back panel, though we were unsuccessful in our attempts to track notes from a MIDI-equipped keyboard; maybe that manual would have come in handy after all! The very fast access time made drum triggering a breeze, so we ended up triggering additional drum parts with the thought that we might use them.

The Eventide SP2016 Effects Processor/Reverb truly is the next step. To my mind, it offers more creativity for your effects dollar than any unit I have ever used, including a 1.6-second looping capacity at a 16-kHz bandwith in the Loop Edit mode (or 3.2 seconds at half bandwidth). To locate the Loop Edit mode, you simply scroll through the available programs, such as Stereo Room, Room Reverb, Chorus, Multitap Delay, Musical Combs, etc., via the Program button, press the Execute button and you are locked

**The Eventide SP2016 can be triggered manually, or via an external audio signal.**

into that program. To load and store the desired sound sample, you must press the Soft Key function at the precise time your source material is played. You next press the Parameter button, and edit the sample with the Length and Position categories via the adjacent slider. Your sample is then ready to be flown back onto the multitrack tape for storage.

Manual triggering is accomplished by pressing the Soft Key function, while electronic triggering can be effected by patching your trigger source into the SP2016's channel #2 input.

During recording of the Patrick Gammon album, because of lack of tracks, we replaced various drum tracks while mixing directly to two-track.

If stranded in a remote multitrack outpost with only one effects toy at your disposal, it's a safe bet that the Eventide SP2016 would probably fulfill your every sampling — as well as special effects — requirements. ■■■

# DIGITAL SAMPLING DEVICES

## An *R-e/p* Guide to Commercially Available Units

**— AMS DMX15-80S —**

As well as offering two independent channels of delay and pitch-change capability, the Model DMX15-80S can be operated as a single-channel 16-bit sampler (with pitch modification of the sampled sound via the other channel) or, with an optional upgrade, as two independent sampling channels. Standard memory capacity is expandable in 0.408-, 1.6- or 3.2-second increments to a maximum of 2.1 megabytes, while the pitch-change range is two octaves. A built-in Loop Editing System enables manipulation of sampled sounds. The edited sample can be read out as a click-free loop, or triggered from the front panel or via any audio input. Quoted specifications include 90 dB dynamic range, frequency response of -3/+0 dB, 20 Hz to 18 kHz, and less than 0.03% distortion at 1 kHz and full output.

**For additional information circle #210**


**— PUBLISON INFERNAL MACHINE 90 —**

As well as functioning as a two-in/four-out delay, special effects and reverb processor, the Model 90 Stereo Audio Computer can be set up to provide one or two independent (stereo-linkable) channels of sampling capability, with full pitch-shift function on either or both channels. Standard sampling time is five seconds per channel, expandable to a maximum of 21 seconds with additional memory; samples are 16-bit linear at a frequency of 50 kHz. Full editing and looping functions are provided, and samples can be replayed from front-panel commands, a MIDI enable, or via an external audio trigger input. Future software upgrades will enable 64 discrete samples to be stored in the unit, and replayed on MIDI command. Quoted specifications include an output noise of -75 dB at full bandwidth, distortion less than 0.03% at 1 kHz/0 dB, and 20-kHz signal bandwidth.

**For additional information circle #211**


**— EVENTIDE SP2016 —**

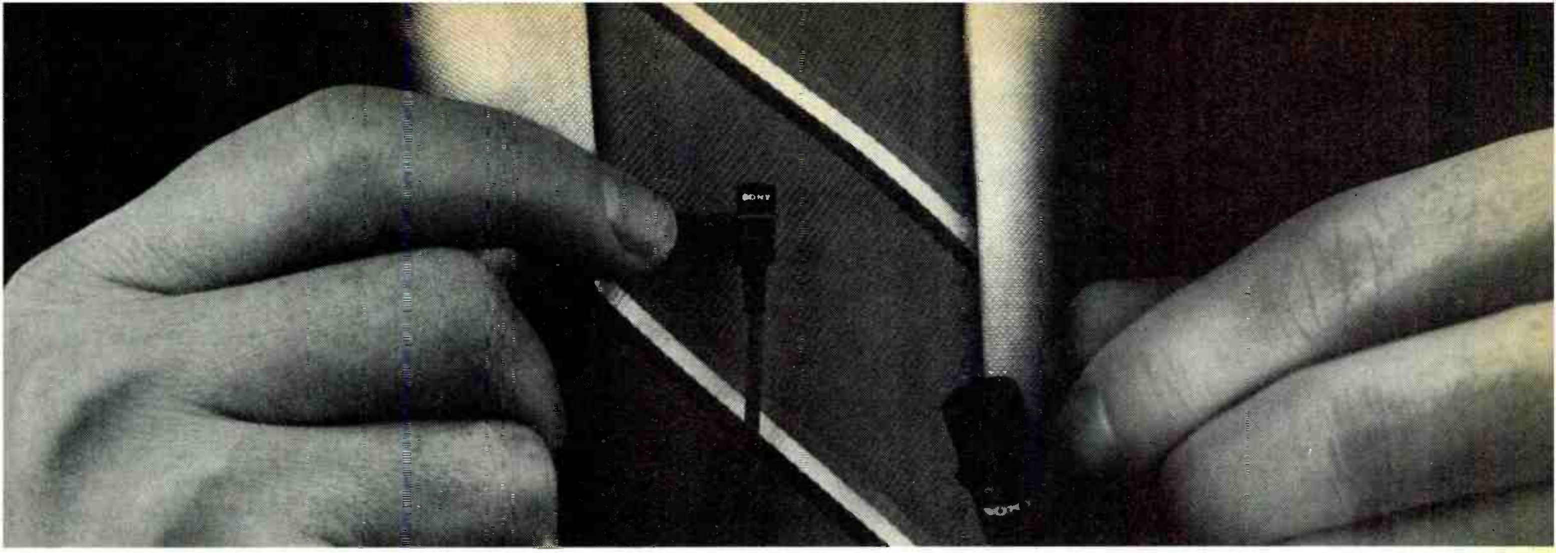
The Model SP2016 Generation II Digital Effects Processor and Reverb features a Loop Edit program that enables editing of sampled sounds on both the front and back segments. The loop or edited segment can be played back continuously, or triggered from a front-panel key or external control-voltage input. Samples are to 16-bit precision, and maximum sample time can be set to either 1.636 seconds at 16-kHz bandwidth, to 3.272 seconds at 8 kHz. The unit can also be remote controlled via an IEE-488 parallel bus. Quoted specifications include a dynamic range of typically 82 dB, maximum THD of 0.1% at 1kHz (typically less than 0.05%), and 20 Hz to 16 kHz frequency range (halved for double sampling capacity).

**For additional information circle #212**

**— OBERHEIM DMS —**

The Digital MIDI Sampler/EPROM Programmer utilizes an eight-bit companding format (plus linear programming mode) to provide a maximum sampling time of 5.33 seconds; the unit's memory can be subdivided so that as many as 16 discrete sounds at four different sampling frequencies — 12, 16, 24 and 32 kHz — can be stored simultaneously. Sound can be edited, reversed, mixed, and overdubbed, and then replayed on command from an external MIDI-equipped keyboard. Play modes include normal, continuous loop, gated and gated loop (with adjustable loop point). In addition, the unit can be used to program EPROM chips for use in Oberheim DX and DMX drum machines, plus others, and will also replay sounds already programmed into EPROMs. The EPROM programming feature is not just restricted to sampled and manipulated sounds; drum sequence programs, synthesizer patches and other MIDI data can be loaded into an EPROM.

**For additional information circle #213**

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**For additional information circle #214**

# CREATIVE AND TECHNICAL ISSUES AFFECTING BROADCAST PRODUCTION AND AUDIO-FOR-VIDEO

## Spotlighting Specialized APfB Facilities

by Ralph Jones

The theme of the recent SMPTE Technical Conference, held in Los Angeles during late October, was "New Directions in Technology — Difficult Choices," a title that aptly describes the current state of broadcast audio and audio-for-video markets. Now that Stereo TV is a practical reality, the video industry has been forced to take a hard look at audio quality standards. The technical and creative implications of this long-neglected issue are far reaching, and the current effort on the part of video professionals to come to grips with the world of audio promises to spur fundamental changes in both facilities and working practices.

With the intention of discovering how these changes might be affecting audio post-production facilities, *R-e/p* visited The Post Group, one of the leading video post houses in Los Angeles. Located in the heart of Hollywood, and already renowned for its video services, The Post Group recently opened a new, "state-of-the-art" audio sweetening facility featuring acoustical design by Jeff Cooper Architects, A.I.A. Companion sidebars to this article will consider the audio-for-broadcast emphasis being placed at Group IV, Hollywood; Post Logic Studios, Los Angeles; and National Recording, New York City.

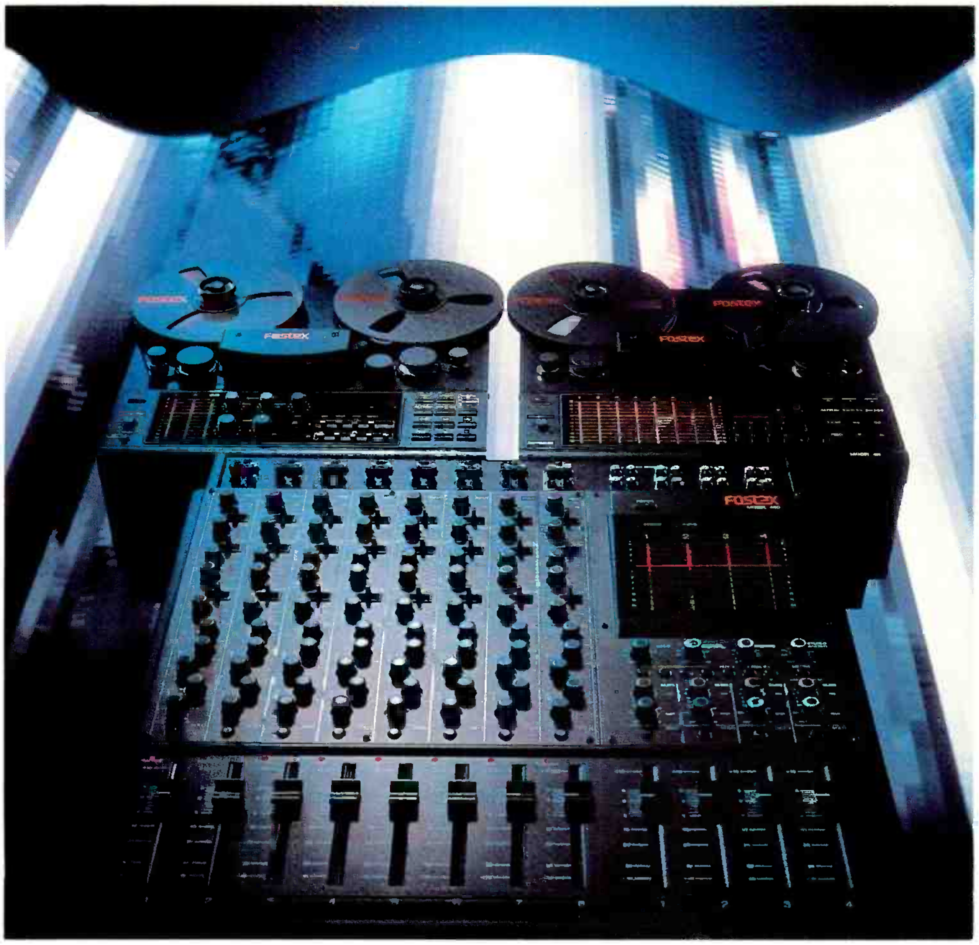
According to the facility's VP of engineering, Rich Thorne, The Post Group "specializes in video effects, and we're trying to develop ourselves in the effects area of audio. "Now that we've opened Post Group Sound," he continues, "virtually every project that we do in video also passes through the audio department." Thorne relates that the decision to branch into audio post was based upon a desire to become a full-service facility. "We wanted clients to be able to come here and, from start to finish, complete a project at the level that we do things. This way, we can provide consistent quality throughout the production."

Housed in a newly-constructed two-story building, Post Group Sound comprises a pair of audio suites, a voice-over booth, and a single shared machine room. Studio A, which measures 19 by 25 feet, functions as the main mixing room, and features a 48-input Neve Model 8128 console fitted with NECAM 96 servo-fader automation, complemented by a host of outboard equipment and large-screen video projection. The main audio monitors are UREI Model 813B Time-Align units; surround speakers comprise a pair of UREI Model 809s in the rear and Boston Acoustics Model 150s at the sides, for a total of six monitor channels.

Studio B, which measures 12 by 16 feet, was conceived as a prelay or layback room, auxiliary to the main studio. The console is a 28-in/26-out Neotek Elite. Both rooms share a common machine room equipped with Otari MTR-90 24-tracks plus MTR-20 four-tracks and two-tracks (the latter configured for center-track timecode), and each employs a CMX 340X video editing system for ATR and VTR machine control.

Thorne has retained the services of two seasoned professional mixers, Peter Cole and Tamara Johnson, for the new sweetening department. Cole and Johnson have followed parallel paths in audio-for-media over the past 10 years, first gaining experience in radio, then moving into television post. (Interestingly, each initially was aiming for a career in record production before becoming seduced by the television industry.) The two collaborated in the design of the Post Group's new audio facility with Phil Mendelson, who now serves as chief technical engineer. . . . continued overleaf —

## THE NEXT GENERATION. THE NEXT STEP.



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For additional information circle #185

## *Enhancing the Quality of* Broadcast Audio

### Goals Of Audio Post Production

Peter Cole and Tamara Johnson are called upon to handle virtually all categories of video product, from in-house sales motivation tapes (referred to as "industrials"), to major network specials. Their work thus encompasses a broad range of audio requirements, from the extraordinarily simple to the relatively complex. In all cases, however, the focus of their efforts is, for the most part, different from that of a music recording engineer.

"The most important factor in our work is realism," Johnson relates. "For example, about half of one television show that I do takes place on a stoop: it's shot on a stage, of course, but it's supposed to be *outdoors*. It's my job to use Foley and effects to heighten that illusion.

Johnson most relishes working on documentaries, since they often require a prodigious creative effort to achieve realism. "I did a documentary

— Tamara Johnson —

on John F. Kennedy in 1983," she relates, "and it was great! The Kennedy family had a historian who shot a lot of footage of the family in 35mm, beginning from their childhood. The prints were beautiful, but they were silent, so I had to create all the sound effects, then add the music and narration."

In projects that utilize narration, the announcer's words are of primary importance, of course, and must not be buried in the mix. "You really want to 'sell' the announcer, because he's the most important thing," Johnson says. "I very often use the Aphex Aural Exciter, because it just pulls the voice out of the mix. I used to do an older show in mono, and the announcer — you could fall asleep listening to this guy narrate! I put his voice through the Aphex, and it brightened him up. You were more apt to listen to him; he caught your ear."

Another important factor in achieving realism is the technique of ambience regeneration. For example, when dialog is looped during ADR sessions, the new lines are usually recorded in a studio with little or no room ambience. But if the original dialog tracks incorporate any degree of room ambience, and the looped dialog sounds dry, the contrast is immediately apparent to the ear, resulting in a synthetic, unnatural quality. It is the mixer's job, in this case, to add matching ambience to the looped dialog tracks at mix time.

"The Quantec Room Simulator has bailed me out many times," Cole offers. "For example, I recently had to do some loop lines where we weren't replacing the entire sentence, but only certain words. I had to match the ambience of the original dialog, which was recorded in a boxey-sounding living room, and I was able to do it *very* convincingly with the Quantec. That kind of outboard gear is *not* a luxury

— Peter Cole —

## AUDIO REQUIREMENTS FOR SCORING:
### A Conversation with Dennis Sands, chief engineer, Group IV Recording, Hollywood

*by Ralph Jones*

Since Group IV Recording first opened its doors in 1976, the facility has earned an enviable reputation as one of the top scoring studios in Hollywood. Serving both the television and feature-film communities, Group IV has played host to a virtual "Who's Who" of contemporary composer/conductors. The facility's credit list includes such popular network productions as *Hill Street Blues*, *Cheers* and *St. Elsewhere*; television specials and telefilms the likes of *Kenney Rogers and Dolly Parton Christmas*, *Fatal Vision*, *Kent State* and *Children In The Crossfire*; and several major motion pictures, including *Tootsie*, *On Golden Pond*, *Back To The Future*, *Romancing The Stone*, and *Rocky II*.

Group IV features a large scoring area capable of accommodating a 60-piece orchestra, and a spacious control room, the centerpiece of which is a 32-input Trident A-Range console. The audio monitors are tri-amped Sierra/TAD units, with JBL Model 4311s for the surround channels. The studio tracks to Studer A-800 and MCI JH-114 multitracks, mastering to Studer A-80 and MCI JH-110 two- and four-tracks. A separate machine room houses a complete 35mm mag chain, with Magna-Tech four- and six-track film recorders and reproducers, as well as a newly-installed Telecine chain. Clients may score either to ¾-inch video or, in the classic manner, to 35mm workprint projection.

*R-e/p* caught up with Dennis Sands — Group IV's chief engineer — between scoring sessions for the feature film *Clan Of The Cave Bear*, and asked him about the audio requirements for scoring, beginning with the differences between scores intended for teleproductions and feature films.

"It's not that easy to separate them anymore," Sands offers. "In a way, video is often simply the marketing medium, since the product actually is shot on film.

"Interestingly enough, in feature-film scoring, we seem to be working more and more with video, and less with film. In most episodic TV work though, we still work to film for the most part, and that's often simply because they have no time to transfer to video: sometimes, they practically bring the film over here when it's still wet from processing!"

"Right now, although our primary emphasis is on scoring, we're in the process of expanding our capabilities, and we'll be doing total electronic post-production starting the beginning of next year. In preparation for that move, we've put in our own Telecine chain, so that we can make high-quality film transfers to videotape with timecode. As you know, film and video have different frame rates. To reflect this timing discrepancy, the film is actually slowed down slightly when you make a film-to-video transfer. This is done by resolving to 59.94 Hz, rather than 60 Hz. That way, when they dub to film, and everything is slightly sped up again, you're still in sync. But if you're just locking to video timecode and you're not resolving to that timecode's sync pulse when you make the transfer, the music will be out of sync when it's dubbed to film.

*. . . continued overleaf —*

# SCORING FOR TV AND FILM — *continued . . .*

"Precise synchronization is *very* important in scoring. Music has very critical requirements for sync, and can be very difficult to deal with if you have to correct it for timing; dialog and effects are easier in this regard, since you have much shorter pieces of information.

"The music has to fit the picture, after all. All the stings have to be right on the picture cuts: it's all very mathematical, and calculated out to the frame. So, you can understand that if I don't have the proper sync reference for my transfer, then when they take it over to the stage and try to dub to film, the fact that it was in sync with the video doesn't mean anything. That's why [maintaining the correct timing relationships during the] telecine process is so important."

What methods do you use to ensure correct audio/video synchronization during scoring, we asked?

"We drive our system here with our multitrack — having transferred SMPTE timecode onto it from the videotape — and we synchronize using the Cipher Digital BTX Softouch system. This gives us two benefits: first of all, your master should be the 24-track — that is what your [audio] product is on; and second, when the video slaves [to the multitrack] you don't have to wait for it. You're on-line, and you deal with the 24-track like any other 24-track. You can forget about the video deck, and just do your work: everything else follows along. It's fast, and it's easy to do."

How much of the teleproduction work that passes through Group IV is recorded for Stereo TV broadcast?

"NBC is now committed to stereo broadcast, and we do a number of shows for them. *Cheers*, for example, has gone stereo this season. We score that show, and everything, including the title theme, has been converted to stereo. *Entertainment Tonight* is also broadcast on NBC, and the same is true for that show: all the music is being remixed in stereo. *America* is stereo. And we've also done a number of *Movies Of The Week* in stereo.

"Since *Cheers* is shot and posted on film, it goes out on three-stripe mag, but *Entertainment Tonight* is not: it goes out on quarter-inch two-track — with no center-track timecode, because the music is rolled in wild."

What are the basic criteria that guide you in mixing a scoring session?

"What's important for me, as a mixer, is to be able to *feel* the music: understand what it's doing, and what is a proper blend. Obviously, with a record, you're dealing only with music, and you mix a little differently; with music for picture, you have to consider all the other elements — the dialog, effects, Foley, and so on. You might not want that kick drum as hot as you'd have it for a record: you want it there, but you know it's going to be played down, and you don't want it to get in the way of something else.

"Quite often, we will record with a rough dialog and effects track played back as a reference. Then, you know what has to be heard: Is there anything sticking out or anything getting in the way? It gives the composer an excellent reference, too, because he may have written a specific cue with the picture a certain way, and they may in the meantime have

*Above:* **The Scoring Stage at Group IV, Hollywood.**
*Left:* **At the Trident A-Range console (*L-to-R*), owner Angel Ballister, chief engineer Dennis Sands, and composer Alan Silvestri during a music scoring date for the film *Clan of the Cave Bear*.**

**— Phil Mendelson —**

— it's really necessary."

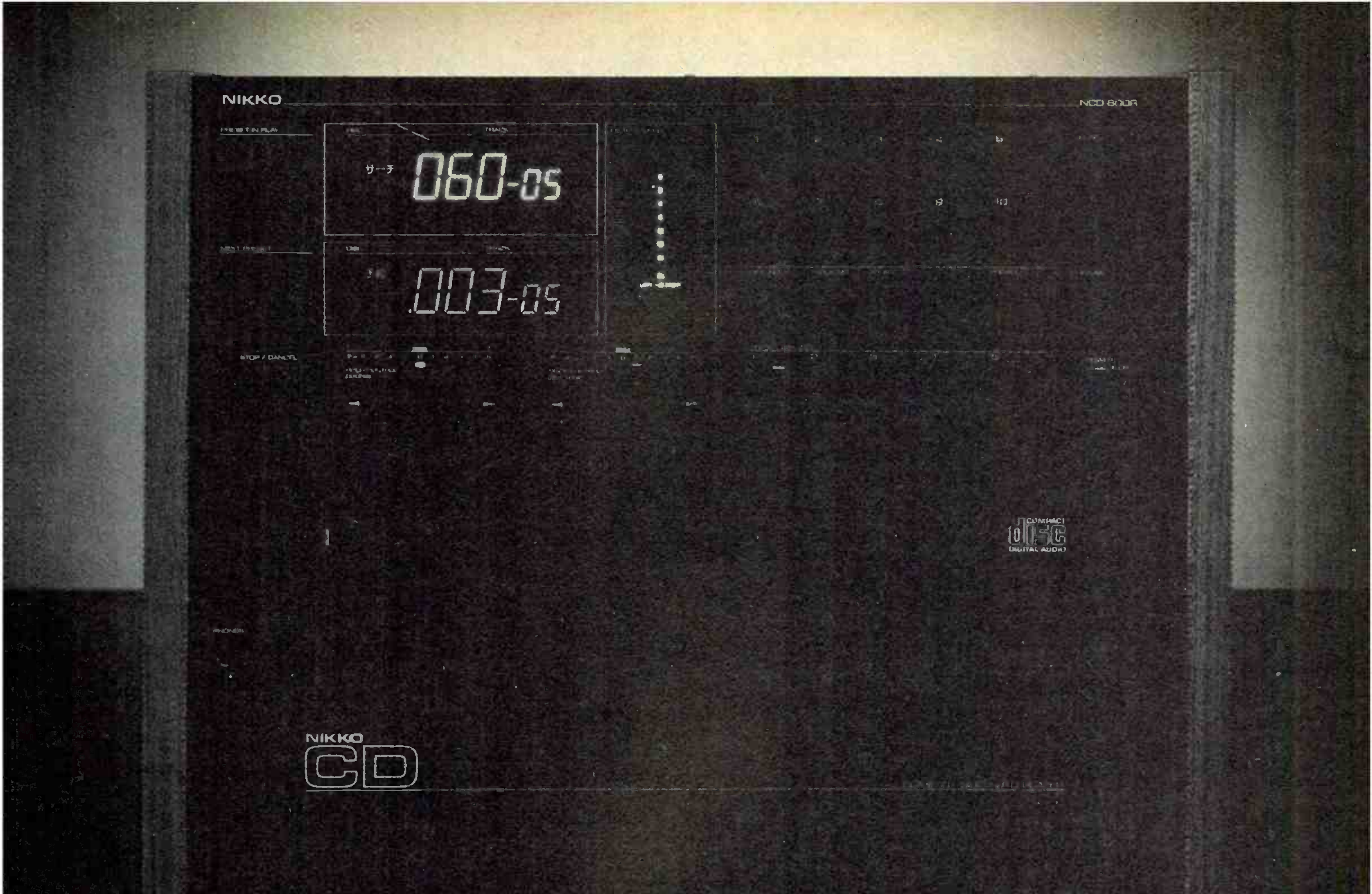
A related technique for heightening audio realism is the use of what is referred to as "room tone," or recorded ambient background noise. When added to the mix in the absence of music, room tone can subtly enhance the realism of a scene. Cole and Johnson maintain a collection of room tones stored in loop form. "But if you really want to do it right," Cole offers, "you find a piece of room tone from the production audio track and use that. If you've got two seconds or so of room tone, you can dub it a number of times to quarter-inch, and make a loop. Lately, however, what I've been doing is to lock a sample into the AMS [DMX15-80S stereo delay unit], trim the loop until it sounds good, and then play it out as long as I need to."

"All these techniques are our 'smoke and mirrors'," Johnson concludes. "In a way, our work is all lies. Once, a lady on a tour through the facility actually attacked me, because she found out that I put laugh tracks on a show. Was she upset!"

### Mixing For Stereo Television

Certainly, the principal focus for current discussions of audio post-production requirements is the question of stereo audio for broadcast television. While neither ABC nor CBS currently provides stereo programming, some affiliates of each of these networks are already on-line in stereo, and NBC is both providing stereo programs and very visibly promoting stereo audio. In this context, one might reasonably expect that facilities such as the Post Group would be making extensive use of stereo audio.

Tamara Johnson currently provides post-production on four sitcoms for

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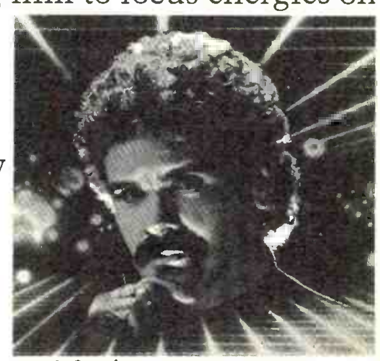

*Frank Serafine—Motion Picture Sound Designer/Musician Credits: Tron, Star Trek I and III, Brainstorm, Ice Pirates*

**For additional information circle #187**

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The Post Group's new Studio A features a Neve 8128 console with NECAM 96 automation, seen here with the Otari MTR-90 remote and CMX synchronization controller.

network broadcast: *227, Different Strokes, Facts of Life* and *Who's the Boss?* Both *227* and *Facts of Life* are produced for the NBC network, and employ stereo audio — to a degree. "I'm finding that I use pan very conservatively," she reveals. "Instead of panning things wide, I pan them quite narrow. What I'm really looking for is a little bit of depth, rather than a big spread. Even when something's happening off-camera on the right side, I still only have it at about 2 o'clock."

Johnson's conservatism stems in part from her experiences as a viewer: she receives stereo television in her own home. "I still find it very strange hearing stereo television," she relates. "It's very different. You know, most people are like me: they don't have a large screen at home. So, peripherally, it's a very narrow angle that their vision is seeing, and to hear a wide aural image with this tiny screen is, I think, very disconcerting."

More significantly, Johnson also is responding to even greater conservatism on the part of producers of network programs. The reasons for this attitude are no doubt complex, but two predictable factors stand out: time and budget. "I'm doing a television show Friday morning, and it airs Saturday," she relates. "I don't have a lot of time to play with it: they have no time even to have music! Creativity is often stifled because of that.

"Last week, I had a show that was supposed to be taking place outside. All the sound effects were in stereo: stereo traffic and ambience, and a helicopter came over from behind and flew by — all in sound effects, of course, since it didn't happen on camera. I literally had one pass to mix it in, and the producers were saying, 'Oh, that's fine, let's move on!'

"In television post," Johnson continues, "the situation is emphatically the *opposite* of that in theatrical sound. I just did a sitcom that was totally underscored — which just doesn't happen very often. But they wouldn't let me do the sound effects in stereo, because of the time that it would take. The only thing that they want in stereo is the music, and that results in a *very* strange sense of perspective. Why should the source music from the hi-fi in the room be in stereo, and everything else be in mono?"

Unfortunately, whether or not it makes sense sonically, "stereo" shows for network broadcast or syndication are very often mixed exactly as Johnson describes: stereo music, with mono effects, Foley, and dialog. But, regardless of whether it stems from time pressures, budget considerations, or a mistrust of current home entertainment systems, the strategy of mixing in this manner may well backfire on television producers.

In a recent issue of *TV Technology*, Jim Swick of WTTW Chicago told of his experiences with the use of a stereo synthesizer to process monophonic programs for stereo broadcast (a practice that is universal among sta-

## SCORING FOR TV AND FILM — *continued . . .*

re-edited [the picture], or added an effect or something.

"You also have to be careful with things like delay and reverberation because, in certain cases, you can't make *too* much out of the music. You can't make it too broad, or thick, or heavy, because you've got these other elements that must come across. If need be, they will be made to come across at the expense of the music.

"And you have to be able to do all this live: in television work, because of the time pressures involved, all the mixing is done 'on the fly.' So you've got to know your room, what the program is, what the composer wants — and you've got to understand the medium.

"In many ways, stereo is easier to mix than mono, because of the 'space' you have: you've got a little more latitude as a mixer for separating things and making them more audible on their own — more room for instruments, and for the interesting sonic things you can do to make the music fit with the picture.

"But, for some time to come, you still have to deal with mono compatibility. You have to be aware of what you're separating: what will happen if one channel is gone, and how you can protect both yourself and the product. You've got to look at the worst-case scenario, rather than the ideal situation. If your product works well in mono, then it will definitely work in stereo — but not necessarily *vice versa*."

Now that Stereo TV is a reality, are you seeing any change in the attitudes and expectations of producers with regard to stereo production, we asked?

"I think it's a little early to see a major change out of production in general. There are some very aware producers, but I think most people are still watching to see what's going to happen.

"Remember, they're up against the wall just doing what they're doing now! They work against very tight deadlines, and when you start to introduce something that's brand new — and nobody is quite sure yet what stereo entails — I can't blame them for being leery. After all, how many homes have you been in where one speaker is on a bookshelf, and the other is hidden behind an ottoman, holding a potted plant? In situations like that, any dialog that's not panned center may be lost. And if I were a producer, I'd make sure that the audience will hear the words, no matter what.

"People in general are receptive to high-quality audio for television, and I think that, ultimately, it'll have the same impact as Dolby Stereo had in film. Great soundtracks have done so much to enhance pictures, and I don't see any reason why they wouldn't do the same thing in television. There are so many creative things you can do with audio to enhance an audio-visual product. It makes no more sense to ignore audio than to ignore video. The people in teleproduction who will succeed, and be at the forefront in 10 years, will be the ones who receive the new technology with open arms. It can only improve the product, and make all of us prouder of what we're working on." □□□

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tions that currently broadcast stereo). Swick allowed that prolonged use of the synthesizer could cause problems, because it results in a more spacious soundfield than that presented by most "true stereo" programs. According to Swick, when WTTW airs actual "stereo" programs, the station often receives calls from viewers complaining that the stereo is inferior to that on the rest of their programming!

For their part, Johnson and Cole are eager to do more sophisticated stereo mixing for broadcast, and would like to see their efforts matched by increased sophistication in production. "It might be interesting," Cole remarks, "to record dialog in a different way, instead of using the normal boom mikes and so on. For example, you could use the [Calrec Soundfield] Ambisonics system, so that when you get into post you could steer the [audio] perspective."

In the hope of encouraging experimentation with stereo audio techniques, the Post Group recently completed a promo of stereo-audio capabilites for the NBC network. The piece, which also features special effects produced by the company's video facility, will be made available to all NBC affiliates that are currently online in stereo. "We put music and sound effects to it," Cole enthuses, "and were able to produce and mix it ourselves when nobody was around!" One can only hope that their efforts will result in increased awareness among television producers of the power of high-quality stereo audio.

### Technical Limitations On Audio Quality

When a project goes through audio and video post-production at the Post Group, it most commonly arrives and leaves on one-inch videotape. The audio may or may not be stereo: chief technical engineer Phil Mendelson estimates that, at this writing, approximately 30% of all Post Group projects employ stereo audio. Regardless of whether it is stereo or not, however, the audio quality of the finished product is limited by a number of technical factors.

One such factor is the inherent limitation of the one-inch video format's audio tracks. "One-inch can sound pretty good," Mendelson observes. "We have a brand new Sony BVH-2000, and I keep it tweaked so it sounds as good as it can get. But the tape is moving at 9.6 ips, is not really optimized for audio to begin with, and has a *very* thin substrate."

Sometimes, audio quality problems arising from the limitations of the one-inch format are aggravated by poor recording practices during a project's production phase. After all, Cole and Johnson must generally work from audio tracks that arrive with the video, and these originate in a situation where sound quality is beyond their direct control. While the picture quality of incoming projects generally is consistent, audio quality varies radically — due in part to a lack of standard practices for maintaining the audio section of production recorders. Cole and Johnson maintain that they encounter frequency-response aberrations and azimuth errors relatively frequently. "Sometimes, we're lucky to have any high end left on the tape!" Cole observes.

Audio quality is further limited by the generation loss associated with the post-production process. "The original production audio is first bounced onto another master, so it's two generations down when it arrives here," Cole explains. "We lay it down to the 24-track, so then it's three generations down. When we mix it, it goes down another generation. Then, we lay it back to the one-inch edited master: that's a total of five generations. Sometimes, the tape has been edited

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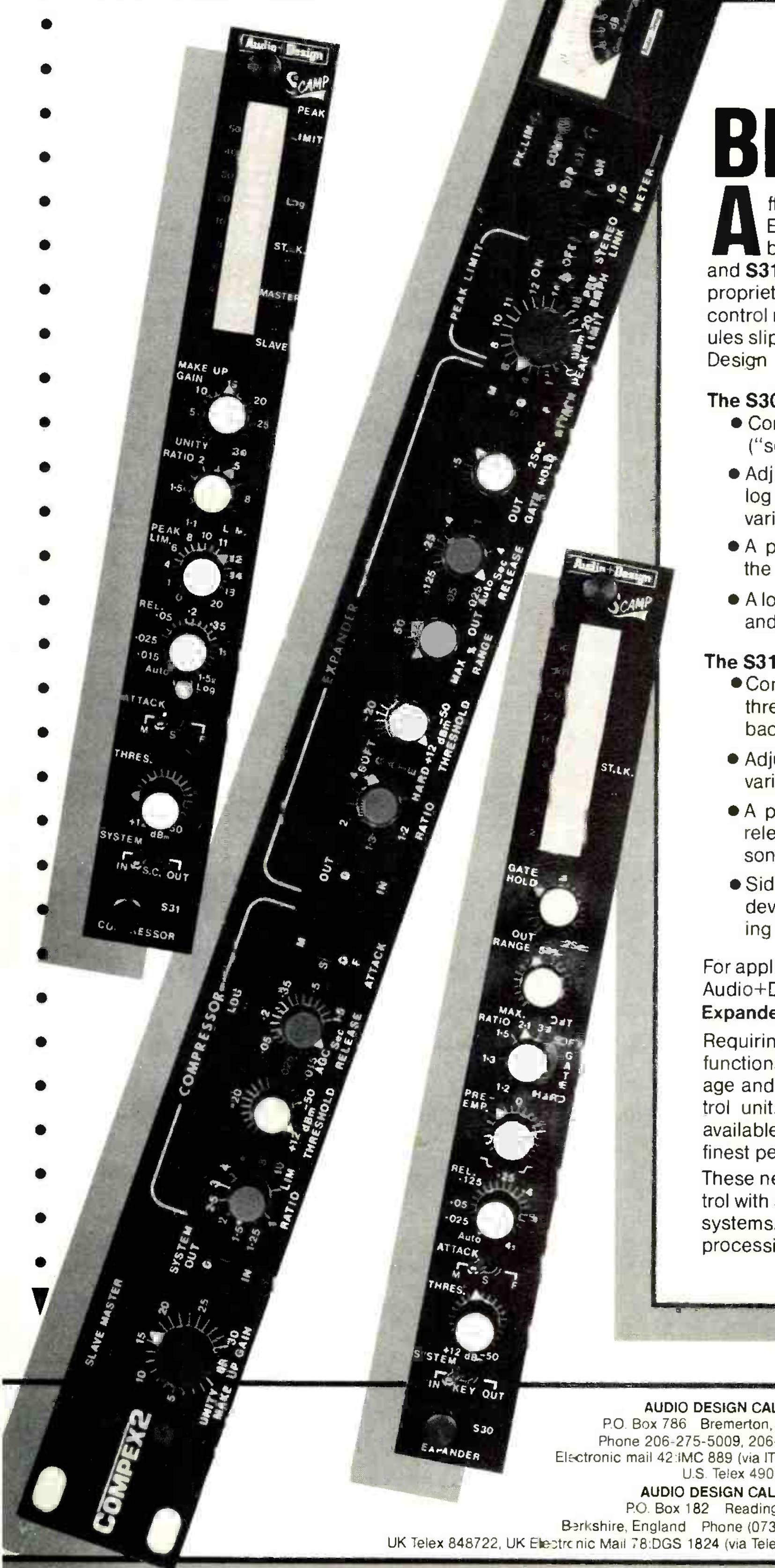


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and re-edited before we even get it, and often the sound was recorded at the wrong level to begin with!

"You know," Cole observes, "it's really nice when someone lets you rebuild all the audio tracks, using the one-inch audio only as a scratch track. That's not at all the common practice with our routine day-to-day work, though."

Johnson also admits to some frustration with the audio quality of one-inch machines. "Why can't they put digital audio on one-inch?" she laments. "It's really silly that the audio on our mastering machine is inferior to what the consumers can record; a Beta Hi-Fi machine is much better!" [It should be noted that the proposed SMPTE/EBU format for digital videotape recorders includes the provision of four PCM-encoded audio tracks. Also, there are rumors that several VTR manufacturers are developing techniques for simultaneously recording a full bandwidth NTSC color video signal *and* two channels of 16-bit/44.1-kHz digital audio on standard one-inch and U-Matic decks — *Editor*.]

The Post Group has placed an order with The Droid Works for a SoundDroid digital random-access editing and mixing system. Mendelson sees the device as a potential solution for, at the least, the generation-loss problem. "For example," he offers, "an editing client could come in with all his one-inch production masters and immediately dump all the audio into SoundDroid, with timecode. Essentially, you could have one optical disk devoted to a program — [recorded] in digital form, and only one generation down. You can do whatever you want with the video editing: off-line edit on ¾-inch, on-line edit, whatever.

"Then, once you've completed your on-line video edit, you give the edit decision list to SoundDroid, which online edits the audio in the digital domain.

"At that stage," he continues, "you've gone no more generations, and you can do one of two things: you can lay back to the one-inch and have a finished, edited master; or, if the audio needs to be sweetened, you can lay it onto an analog multitrack — or in digital form to a digital multitrack — and sweeten from there."

SoundDroid should also be of immense value in dealing with sound effects, since it offers random access to disk-based digital audio tracks. For the present, however, the Post Group maintains a sound-effects library in two forms: quarter-inch audiotape with center-track timecode, and 15 ips stereo audio carts for the most often-used effects. The timecode locations of effects stored on quarter-inch format are logged in a computer database. "We can go to the computer, dial up an effect, and it will tell us its timecode location," Cole explains. "Then, you just type that number into the CMX, and you've got it. So, it's the next best thing to random access."

### The Music Industry And Video Post

Many of the projects that come through the Post Group rely heavily on music tracks, and both Cole and Johnson have suggestions to music engineers for procedures that could

## APfB FACILITY SPOTLIGHT: NATIONAL RECORDING, NEW YORK CITY

*by Rob Tuffly*

National Recording first began life in 1959 as an audio-production facility, then moved into video production and post-producion in 1973, before recently adding a third audio-for-video sweetening suite, and redesigning its main recording studio. Currently, the facility consists of six computerized video-editing suites: two inter-format rooms for Betacam half-, ¾- or one-inch videotape editing; and four rooms for one-inch editing. The complex also includes a film-to-tape transfer and color-correction suite, and two video production stages. The video suites are complemented by three audio-for-video sweetening rooms, one referred to as the "Vidimag" room, which offers double system "film-style" mixing for those clients who are accustomed to editing audio in a sprocketed medium. The two remaining 24-track sweetening rooms, called "Q.Lock" rooms, feature Sony MCI multitracks and consoles, in addition to the synchronizer unit for which the rooms have been named: an Audio Kinetics Q.Lock 3.10. National also has five audio voice-record studios for taped radio sessions.

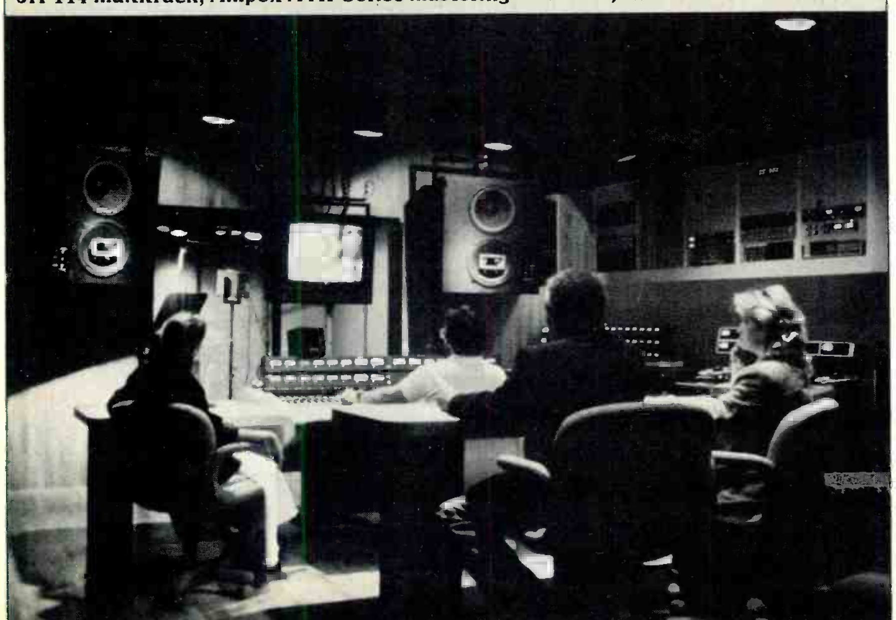
National's effects library consists of 50,000 different sounds on tape, and a Kurzweil K250 digital music system (owned and operated by independent musician/synthesist Norman Casow) is used to generate and create new sound effects. In addition, the studio itself owns a Roland Juno 106 and Yamaha DX-7 synthesizers, and an ARP keyboard array.

According to Irving Kaufman, co-owner of National Recording with Hal Lustig, the facility has existed for 20 years, with in-house production ranging from *Dick Cavett Show* broadcasts, Dr. Ruth Westheimer's *Good Sex* series, to numerous album projects, Music Videos, commercial post production, film scoring and re-recording.

The main complex is located in the former Westside Airline Terminal — the entire four-story National Recording complex stretches for one square block. Recalling the origins of National Recording, Kaufman says: "We constructed the complex with audio compatibility as the key. Our clients can record in one room and mix in another without any problems."

Interestingly, the main audio suite, called Edison Hall, is located in the Edison Hotel some blocks away from the main complex on West 42nd Street. "Our business is constantly evolving," says marketing manager Bill Kelly. "We previously had a 12,000 square-foot facility on Fifth Avenue before our move to 42nd Street. Because of the special require-

**National's latest audio-for-video room houses a Sony MCI JH-500 automated console, JH-114 multitrack, Ampex ATR-Series mastering machines, and UREI 813B monitors.**



make the audio post process easier and result in a better end product. For example, concert performances have proven to be very popular in the Home Video market, and many such programs are post-produced at the Post Group. Concert videos rely in part on the artist's patter between songs — both to maintain continuity and to give the viewer a sense of "being there." But music mixers are not always sensitive to the importance of this extra-musical material.

"Music mixers are only thinking about the music itself," Cole explains. "But I need them to process all the dialog in the same way as they do the vocals. They'll mix to the end of a song, and then maybe just 15 seconds into the dialog. As a result, the dialog may be on the 24-track, but *not* in the final mix. When this happens, I have to go back and try to process the dialog to sound like what they did to the vocals, so it'll match. This takes a lot of time: matching their processing isn't always easy! I need them to mix straight through, and then come back and do overlaps, so I have the dialog, too."

The search for audio realism also extends to music mixing in concert video post-production, albeit in a more subtle form than in dramatic shows. Here, the task is to work with

## APfB Spotlight: National Recording — *continued . . .*

ments involved in music recording, we leased a large, former ballroom from the Edison Hotel to build a main recording studio."

At the time of writing, Edison Hall is being redesigned by well-known acoustician Tom Hidley, with construction contracted by Sierra Audio. According to Kelly, after its redesign, the "showcase" studio will measure 22 by 50 by 40 feet (H×W×D), with a spacious control room measuring 25 by 30 feet. Equipment upgrades will include installation of a new 56-in/48-out Solid State Logic SL 6000E Video System linked to an Otari MTR-90 24-track and Ampex ATR-100 two- and four-track machines. Timecode synchronization and control for the audio and video transports will be provided by an Adams Smith Series 2600 system. Outboard gear is scheduled to include compressors from UREI and dbx; Valley People Kepex II noise gates; Dolby noise reduction; an AMS RMX16 digital reverb; and UREI graphic equalizers. Crown and Bryston amplifiers will drive UREI 813B audio monitors.

Having come full circle from strictly audio production, before moving towards a video production and post-production emphasis, to recently offering high-quality audio coupled with full video production, Kaufman offers that National remains on the forefront of audio-for-video production. And, seemingly, the facility is re-affirming its audio stance by reinforcing its existing audio hardware with newer upgrades. In addition to renovating its Edison Hall studio, National recently added a third audio-for-video studio, designed by Kaufman and chief audio engineer Eldo Luciani.

The new 24-track room houses a 28-input Sony MCI JH-500 automated console, a JH-110 24-track with Dolby NR, and Ampex ATR-100 four- and two-track machines, the latter being interlocked by an Audio Kinetics 3.10 Q.Lock synchronizer. Powered by Crown amplifiers, monitoring is provided by UREI Model 813Bs mounted on adjustable tracks for horizontal (forward to backward) movement to accommodate variable listening positions in the control room. The room has acoustic paneling set into carpet walls, wood accents, and a complete set of bass traps for enhanced acoustic accuracy in the control room.

With the purchase of new state-of-the-art equipment, and the installation of an entire new room, National Recording states that it still places a strong emphasis on high-quality audio for its film and video industries. "All our studios are designed for the client," concludes Kaufman. "We wanted a complex that gives them everything they want in one location." □□□

Outboard equipment in the Post Group's Studio A includes a Lexicon Super Prime Time, AMS DMX-15S delay, dbx compressor limiters, and a Quantec Room Simulator.

balances in the mix, complementing the visual cues in the image. On a closeup of the drummer, for example, slightly boosting the drum track can enhance the impact of the shot, and thus propel the audio/visual experience forward.

But such decisions have to be made in post-production, after the final video edit is completed, and the artist normally wants their own engineer to mix the music. To accommodate this requirement, while still allowing for some rebalancing in video post, Cole asks music mixers to prepare a special six- or eight-track music master, recorded such that the six or eight channels replayed at unity gain represent their intended music mix. The channels are allocated to separate stereo submixes of various musical elements that might be featured visually in the video: for example, the lead vocalist, drums, lead guitar, and so on. In post, then, Cole has these elements available for featuring if necessary, while still retaining a reference mix of the music track.

"I would like this to became a more-or-less standard method," he offers, "but one of the problems that I've run into is cost. For instance, I have them add all their processing to each stripe. But that means that you need separate [echo] chambers for each mix, so the outboard requirements go way up. You need to keep each stripe pure, after all: you don't want the reverb from the drums going into your vocal track!"

Music Videos might appear to involve similar concerns. In reality, however, they are almost never sweetened: for the most part, the Music Videos done at the Post Group require only video editing. "A lot of times," Cole explains, "it's just a pre-recorded music track that's already on the one-inch." (The exception, of course, is concert videos, which usually are sweetened for sound effects — principally crowd noise and ambience.)

Post Group mixers would like greater creative control in Music Videos, however. "Your eye is telling you what you should be hearing," Johnson observes. "On the record, for instance, things might be panned totally opposite from their positions on the screen; it would be nice to correct things like that."

"Also," Cole elaborates, "in most music mixes, your vocal is way down in the track. But, a lot of times, if you've got a closeup of the singer, and the vocal track is just barely cutting, it comes across wrong. It would be nice to be able to boost the track a little bit."

It seems likely that Cole's technique of preparing a six- or eight-track master for video post-production might also be of use in the production of Music Videos, particularly in the case of "performance" videos. Given an automated console, the actual cost of making the extra masters should be minimal, since all that would be required is to re-run the automation record with different output assignments. The decision to take this extra step is, in part, a creative one, however, and ideally would be made by the music producer in consultation with the label (and, perhaps, the video producer, if available).

### Synchronization Standards

Not surprisingly, both Peter Cole and Tamara Johnson relate that the problems they encounter most frequently in music-related projects have to do with synchronization. "For example, take a project like the *Jane Fonda Workout*. In projects like that, the music sometimes isn't played back on the shooting set at the correct speed [or film/video timebase], and we have to re-sync it all," Johnson explains. "There seems to be widespread misunderstanding about how synchronization should be done. It's very simple if you know what it is, but the people who *don't* understand it can create major problems."

Frequently, music engineers aren't aware of the amount of timecode preroll that is required for video post work. "I often get tapes with two seconds or less of code rolloff at the head," Cole complains.

"Another of my 'favorite' things," adds Johnson, "is to get a master of all the songs with timecode, all very nicely leadered! No preroll at all!" Cole contends that such problems are so common that he doesn't much worry about timecode anymore. "If I don't have timecode, I'll just VSO it in, and I can get it pretty much field-accurate. You can tear your hair out trying to tell people how much preroll you need. 'Fifteen seconds? What do you mean? That's so much!' There ought to be a standard for timecode preroll."

Even when adequate timecode roll-off is provided on the tapes that Cole and Johnson receive, the code itself may not be in the right form. It is not uncommon for them to get tapes with 60-cycle code — the standard for film — rather than the 59.95 Hz code required for video. "People don't seem to understand timecode very well in general," Johnson complains. "I've even had tapes in which they've recorded the music, then gone back later and put timecode on. Well, that's not going to do you any good! It's a general misconception among people who work in the field. They certainly know how to lay their cables correctly so we *don't* get buzz, but they don't have to work with the end result. To them, timecode is just this funny-sounding thing that you put on track four, and that's it. But it's critical!"

### Interfacing Audio Machines With Video Systems

Both of the Post Group sweetening rooms employ a CMX 340X for machine control. While the CMX functions primarily has a video-editing system, standardization on one system for all machine control offers some distinct advantages. "Using the CMX with its distributed synchronization," Phil Mendelson explains, "we can patch in a one-inch machine from the downstairs machine room —control signals, audio and video — and lay down or lay back from either upstairs audio suite. That kind of flexibility is possible because of the combination of CMX compatibility with access to the house routing switcher."

The consequent efficiencies have been gained at some monetary cost, however, since the CMX still requires an external synchronizer to communicate with audio machines. Mendelson explains: "The CMX controls machines by an intelligent interface. The reason it's called 'intelligent' is that the CMX computer has RS-232 output ports to communicate with what is essentially a complete synchronizer: it reads timecode, looks at status from the tape machine, and controls and synchronizes the transport.

"Now, in recent years, most video machines have adopted serial control — including not only motion control, but also control of insert modes, and so on — all through an RS-422 port. The video machine, in turn, reads timecode and puts it on the data bus. CMX has taken advantage of this capability, and made their interface simpler. But, whereas some audio machines have serial ports, the software to communicate with them has not been developed. It's all well and good that my multitrack has a serial port on it, but the CMX interface doesn't know what to do with it. So, what I have to do is communicate serially with a synchronizer. In this case, I'm using a couple of Adams-

## APfB FACILITY SPOTLIGHT: POST LOGIC STUDIOS, HOLLYWOOD

*by Rob Tuffly*

During the last week in October, Post Logic Studios christened its recently opened facilty with audio-for-video sweetening sessions for the NBC Network, and album projects for Island Records. Aimed predominately at the Stereo TV broadcast market, the 24-track facility occupies 3,000 square feet, consisting of one main studio and control room, a Foley area, an ADR (Automatic Dialog Replacement) suite, and a separate machine room.

Before building his new facility, owner Miles Christenson solicited input from potential clients and competitors within the audio-for-video market. "If you talk to people in the industry," he offers, "they tell you that many studios are booked solid for three weeks in advance with top-name clientele. And basically, to book those 'state-of-the-art clients,' you must have 'state-of-the-art' equipment." The following equipment list indicates that Christensen seems to have taken his own advice.

The control room centers around a new 40-input Solid State Logic SL 6000E Video System with eight stereo input modules, Total Recall automation, and Events Controller. For timecode synchronization, an Adams Smith Series 2600 system controls an Otari MTR-90 24-track, an MTR-20 four-track and an MTR-20 two-track. Video decks include Sony BVU-800 U-Matic and a one-inch BVH-2000 VTR. The custom audio monitoring package, provided by Aura Systems, consists of a complex multiway system, configured to include four longitudinally-cut half cylinders, eight 12-inch woofers, four eight-inch mid-range units, and six two-inch HF domes. The system has a quoted frequency response of 20 Hz to 25 kHz, ±3 dB. Referring to outboard gear, Christensen spotlights his two hardest working units: a Publison Infernal Machine digital effects and delay unit, and a Lexicon 224XL digital reverb with version 8.2 software.

When a session requires different musical instrumentation, Christensen drives to his nearby Hermosa Beach studio to pick up any number of his digital synthesizers. Primarily a keyboard-oriented facility, Christensen's 16-track studio, called C&H Productions, houses an E-mu Systems Emulator II, Roland Super Jupiter, Fender Rhodes Dymo-My-Piano, and an original Sequential Circuit Prophet 5. Since both facilities are audibly compatible, Christensen can sometimes begin a project at his ocean-front studio, and then bring the tapes to Post Logic for embellishments and mixing on the SSL.

Christensen's foresight concerning the future of audio-for-video is exemplified by his

Smith Series 2600 systems.

"When CMX finds the time to write the software," Mendelson concludes, "we're not going to need the synchronizer, but you'll still need something at the machine end that reads timecode and puts it on the data bus that will then communicate through the RS-232. Eventually, that will be a function of the audio machine: the video machine manufacturers have been doing it for a number of years."

Studio A's Neve 8128 console is fitted with NECAM 96 automation, which can also be used as a locator/controller for the Otari MTR-90, with the video machine in turn being slaved to the latter. "That's useful when music people come in and don't feel comfortable using CMX," Mendelson offers. "CMX is primarily an editing system, and a lot of people outside of the video industry can't relate to that: it's not the kind of system that you just sit down in front of and know how to use. NECAM acts more like an autolocator."

The NECAM system cannot communicate directly with the CMX, however: it simply slaves to the SMPTE timecode signal from the 24-track. When circumstances arise where both systems must be used, this introduces additional delays into the post process. "It would be nice to deal with a *single* control keyboard, rather than having to communicate with the CMX and the automation separately," Cole observes wistfully.

### Philosophical Differences Between Audio And Video Operations

It is clear from the comments of the professionals working at Post Group Sound that a number of technical and procedural issues must be resolved if increased communication between the audio and video worlds is to be realized. Achieving such resolution is a matter of both equipment design and promulgation of standards. But technical aspects of any given field inevitably affect *and* reflect basic attitudes toward system design and working practices. Given the fundamental differences between the technologies of audio and video production, corresponding differences in attitude may further hinder current efforts at communication and cooperation between the two industries.

Some indication of the philosophical differences between the audio and video fields may be found in the observations of Rich Thorne, The Post Group's VP of engineering. Reflecting on his experience in constructing the facility's new audio sweetening facility, Thorne expresses substantial frustration with audio technology and practice.

"If I could generalize," Thorne observes, "I think that audio is a horrendous industry: it's not been thought out. It amazes me that there is no one manufacturer making enough of anything that you could go with that one company. I think it's a tremendous waste of time, money and technology.

"You can spend a quarter of a million dollars on an audio console that does nothing but switch your monitors and change your levels, and has a computer that makes faders go up and down. Then, you have to spend another quarter of a million on 40 boxes: one cleans up this part of the audio, another cleans up another part of the audio, and yet another distorts the audio! Now, we in video have 'boxes,' but they do very specialized jobs, and we can at least 'talk' to all of them [via serial and parallel computer interfaces].

## APfB Spotlight: Post Logic — *continued . . .*

utilization of the SSL Events Controller to route effects from the studio's pair of Studer A725 Compact Disc players to the mastering medium. In addition, he is looking for a special-effects library recorded on CD rather than tape. "But for right now," he concedes, "we are marrying taped and CD special effects until Sound Ideas [currently one of the companies offering effects mastered on Compact Disc] sends us a complete library on CD. We also work with a floppy-disk library of sounds recorded with the Emulator from a frequently rented [Sony] F1, on which we sample different location sounds. This way, we can keep all our effects in the digital domain."

Furthermore, chief engineer Bill Elswick is currently creating custom software for the IBM AT computer to control the Studer CD players for triggered playback operation. Post Logic's goal is to load up four or five different sequences of sound effects, and reference their playback to timecode recorded on the master tapes.

According to Christensen, Post Logic's main production philosophy coincides directly with its desire for state-of-the-art equipment. "Let's face it, the SSL is the 'Rolls Royce' of consoles. I looked at all the consoles available, and the quickest, most efficient integrated system that includes sychronizers and the ability to run the most transports was the SSL. The Total Recall capability is fabulous; it gives me the ability to run two or three major sessions a day, giving them all the attention that each one deserves. Plus, you can always use it — 24 hours a day.

"I'm going after the networks," he continues. "Before I constructed this studio, I went around asking people that if I built this [kind of] facility, would they give me any work? I soon realized that if you want high-caliber clients, you must have an SSL."

When asked about the current status of Stereo TV, Christensen's reply emphasizes the obvious forward direction of this still fledgling, experimental field. "Stereo TV is here to stay," he states. "The only problem is the last link in the chain: the headroom restrictions of the satellite [audio transponders]. We have to be pretty careful about the dynamic range we release, because the satellites have trouble handling the peaks that well-mixed, dynamic stereo will give. Fortunately, there is a stereo master compressor built into the SSL that smooths everything out."

By electing to use high-caliber audio equipment, locating his studios in Los Angeles (the heart of television and film production), and keeping abreast with the advent of digitally-based hardware, Miles Christensen and Post Logic Studios gives the appearance of a viable, independent audio-for-video sweetening facility aimed specifically at Stereo TV production. □□□

**Post Logic's new facility houses a Solid State Logic SL6000E automated console, and a custom monitoring system designed by Aura Systems.**

"We need to get people out of their garages and get responsible companies building a full piece of equipment — I think it's really going to take the video industry to do that. Video, being a much younger industry, is much more progressive. We solicit change. Audio is more analogous to the film industry: God forbid that something should come along to help those guys! The [audio post-production facility] that we've built is — pardon the expression —'state-of-the-art,' but it's so backward it's ridiculous.

"We purchased a SoundDroid with this problem in mind for delivery early next year, since it has the digital power to handle the kind of software that's going to do different things to sound, along with storage. You're going to be able to put your sound into a single box and edit it, move it around, mix and EQ it, add room conditions — all digitally. I think it's going to be very important to the industry.

"While quality certainly is a factor, what's important to me is that SoundDroid should really save time. Time is money, to be trite, and when you have to do things using conventional analog methods — with very little control over the sound, other than a synchronizer and a splicing block —it's *very* inefficient. The more inefficient something is, the more taxing it is for the creative human, and the less effective he is over a 10-hour session.

"For example, I remember my first experience with a real sweetening session. The guy had razor blades and tape all over the place! After an spending an eight-hour day working on a 30-second commercial, he was exhausted. I was very impressed with how fast he was 'Vegamaticing' his quarter-inch — and it sounded fine — but if we had to do it over again he was in no condition.

"I did a demo recently for a convention, and the audio mixer spent 18 hours sweetening it. To me, it certainly wasn't 18 hours' worth of work: it should have been done in five or six, at most. It's fine to charge people and make money doing a very inefficient job, but I'd like to be one of the ones getting us all out of that. In 1920, you could get from L.A. to New York in a Model T; now you can get there on a plane a whole lot faster. Does that mean the Model T was bad? Of course not, but there are much better ways to do it now, and they're available."

Many of Thorne's criticisms of the audio industry have some validity. It is certainly true that we need to develop more sophisticated (and universally accepted) interface protocols, particularly for machine control. And digital signal processing techniques do, indeed, promise to allow more powerful integration of functions in a single piece of equipment.

But his comments also reveal the extent to which video engineers may fail to understand the legitimate concerns of the audio community. After all, sound is far more subjective than images, a fact that accounts (at least in part) for the proliferation of outboard equipment in audio production. It is highly unlikely that SoundDroid — or any other digital editing and mixing system, for that matter — will prove powerful enough to replicate the subjective effects of every existing signal processor. (In this regard, it is important to note that The Droid Works plan to include analog insert points in SoundDroid's design.)

We've clearly got a long way to go, therefore — not only in terms of our technology, but also in bridging a serious gap in communication — if we are to see high-quality audio make real inroads into the broadcast and video production fields. But audio and video each have unique strengths, and much to teach one another: all that is required is mutual respect and a sincere effort at communication. Given that climate, we may yet arrive at a creative synthesis of the two media, to the benefit of all. In the absence of that attitude, however, very little will be achieved. ■■■


# KLARK-TEKNIK DN780 DIGITAL REVERB AND SPECIAL EFFECTS PROCESSOR

*Reviewed by Bob Hodas*

Klark-Teknik first introduced the DN780 in October 1984 and, in a continuing R&D effort, in May of this year released an updated software version (V1.5). The new software includes brighter reverberation algorithms; improved plate presets; three new special effects programs; assignable remote protection; user-memory protection; and extended diagnostic routines.

As its name implies, the DN780 simulates acoustic environments — such as rooms, halls, chambers and plates — and also provides several special-effects programs. It comes with 39 protected memory locations, currently holding 28 factory-loaded programs, of which 20 can be described as reverb settings, and eight as effects settings. An additional 50 memory locations are open for user storage. The DN780 operates as a mono-in/stereo-out device, and may be operated either balanced or unbalanced.

Having heard this unit only at various trade shows, I was curious to examine the latest version to see if the manufacturer's claims would be supported, considering the DN780's reasonable price tag. As a general introductory statement, I would have to say that the DN780 was surprisingly simple in operation, and capable of creating some very nice ambient spaces.

The unit itself is a 2-U rack mountable unit that comes with an optional remote control on a 25-foot cable. Rear panel connections include an AC power receptacle with a very clever fuse holder — a pull out drawer holding the in-circuit fuse along with a spare. (The fuse may not be removed without first pulling the power cord.) The remote-control connection is also well thought out, with connections to a "Dee"-type socket, easily handling extended barrel screws. This type of connector makes attaching the remote head very easy and secure, since there are no tiny screws to get lost, and no searching for a tiny screwdriver to insure the connection.

Above the remote socket is an empty port for a future computer connection (see description of remote below). The single input and two output connections are standard XLRs, and there are two recessed output-level adjustment screws. A fan keeps the unit cool, although I was surprised to find no intake vents on the chassis. (I guess it draws enough air through the front panel buttons, however, since the chassis never gets very hot to the touch.) Fans are a pet peeve of mine, because they are always adding a few dB of excess noise to the mixing room. Not so with the DN780: this is the quietest fan I have ever heard (or not heard, to be exact); I think I'm going to order one for my Compaq computer!

### Front-Panel Features

Front-panel controls and indicators make operation quite simple. All parameters and memory locations are displayed via easily read, bright half-inch LEDs. When a parameter is selected for adjustment, a small red triangular LED lights up just below the display window. A 10-segment LED input meter is provided, input level being adjustable with a pot mounted next to the meter. A momentary Input Mute switch allows the input to be muted in order to listen to the decay properties of the chosen program. A Reverb Mute switch mutes the unit's output, which may come in handy in cases where low-frequency build up gets out of hand, or regeneration feedback is present.

Seven switches sit beneath their prospective display windows, and are used to actuate the parameter to be adjusted. The parameters are as follows:

*1. Pre-Delay.* Adjustable from zero to 990 milliseconds, this controls the delay time between the initial signal and the onset of reverberation. On certain Reverb Patterns, it is inserted between the early reflections and in reverberation envelope.

*Pattern.* Adjustable in five steps, this changes the number, spacing and density of the first reflections, and is probably the most important parameter to understand, because it will change the complete personality of whatever program you happen to be in.

Pattern #1 is used for all Plate programs, and has percussive, high-density initial reflections building quickly into extremely dense reverberation with smooth, even decay.

Pattern #2 is used for all Room programs, and has high-density initial reflections with the natural colorations found in real rooms. Reverberation build-up is extremely fast, and decay is smooth and even. Pre-delay is inserted between the early reflections and reverberation.

Pattern #3 is used for all Chamber programs, and its percussive, medium-density initial reflections have regular delay spacing, producing a more colored, "live" sound. Reverberation build-up is moderately fast, while decay is smooth and even.

Pattern #4 is used for Hall Programs 1, 2, and 4. Its low-density initial reflections contain discrete echoes simulating the reflections from boundaries found in natural rooms. Reverberation build-up is slow, and decay is initially colored, then smooth and even.

Pattern #5 is used for Hall programs 3 and 5. It is similar to Pattern 4, but has different discrete echoes.

*3. Level.* Adjustable in 10 steps, this determines the balance of the early reflected energy relative to that of the reverberant sound.

*4. Decay.* Adjustable from 0.1 to 99 seconds, this sets the overall reverberation decay time. Upper and lower limits of decay time are determined by the "Room" parameter, as shown in Table 1.

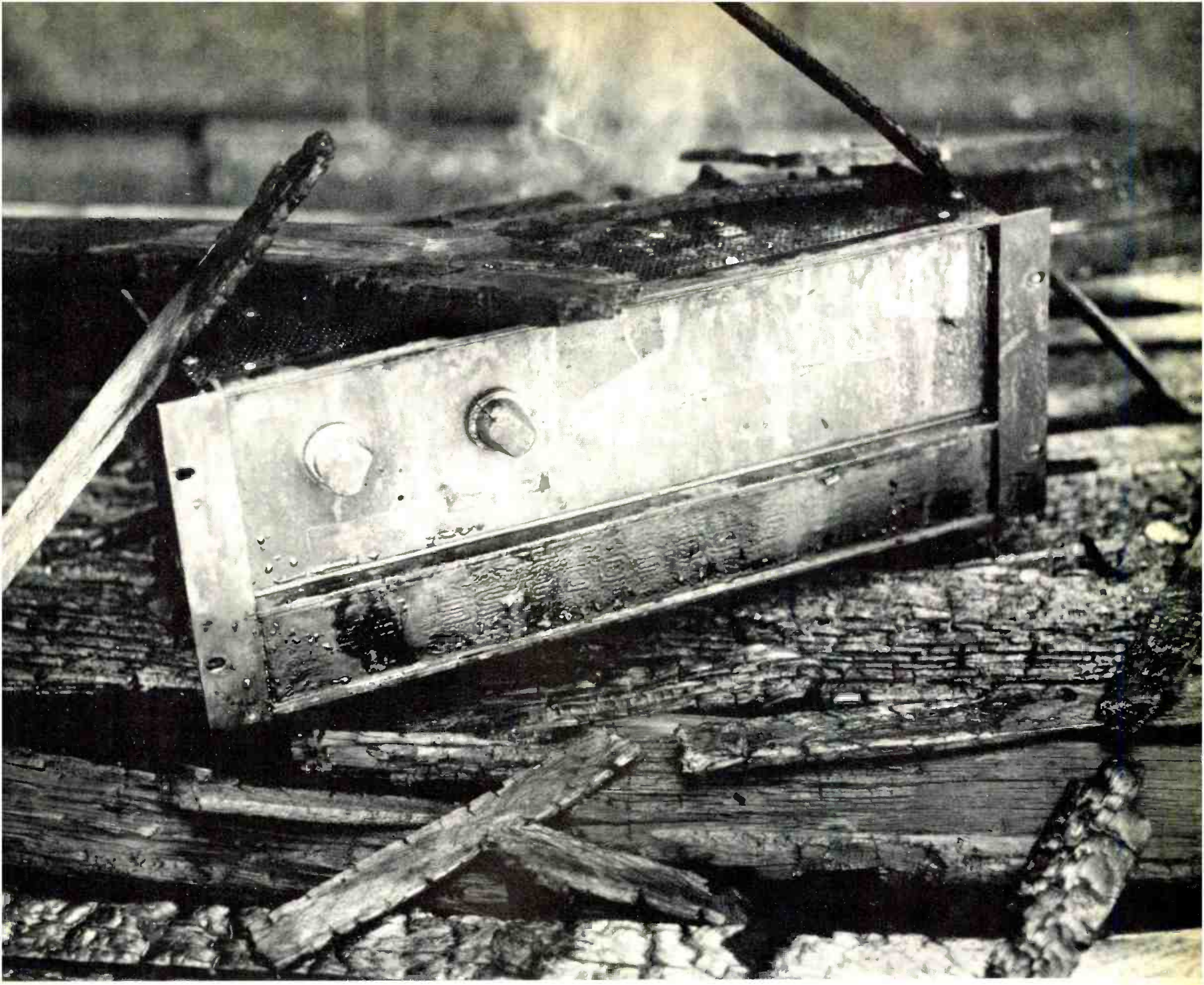
*5. LF.* Adjustable in 15 steps (-7 to +7), this varies the decay time for low-frequency reverberation.

*6. HF.* Adjustable in 15 steps (-7 to +7), this varies the high-frequency reverberation decay time.

*7. Room Size.* Adjustable from 5.0 to 99 linear meters (125 to 1 million cubic meters), this determines the dimension of the simulated space, and sets limits of decay time. Room size is not a factor in the Alive, Gated Reverb or Reverse Decay programs.

All parameters are altered via the

. . . continued overleaf —



Parameter Up/Down keys. Adjustments may be effected in single steps or in slow and fast increments by using key combinations.

A 10-button numeric keypad to the right of the display window is used to call up the different memory locations, and is also used with the *STO*re and *SEQ*uence buttons to perform those specific functions. User-modified program variations can be stored in the 50 user locations through a simple assignment process that also lets you know if you are about to store over a previously filled memory location. Up to 16 memory locations can be stored and recalled in any sequence the user desires. The sequencing procedure is very simple to setup and use, and will tell the user if he is inadvertently storing an empty location in the sequence. Parameters may even be modified for programs within a sequence without having to reconstruct the entire procedure. A foot switch may be connected to the remote connector to allow easy sequence switching during a show.

The Remote Control is small enough to be hand held, and contains Remote Enable and Sequence buttons. Four parameter sliders are normally assigned to Pre-delay, Level, HF and Decay, but can be re-assigned through a procedure using the DN780 front panel.

Each slider operates only when moved to coincide with the current setting for that parameter, and this is confirmed by appropriate Track LED illuminating.

I don't think that the remote is as well thought out as the rest of the unit. Klark-Teknik could have done better with one slider and seven assignment buttons for the different parameters. I felt that only having four parameter adjustments available at a time made this a "partial" remote. The fader reassignment system seemed superfluous, considering that by the time I changed the fader assignments, I could have done all the parameter adjustments right at the unit's front controls.

**Table 1: Decay Limits As Restricted by Room Size**

| Room Size (linear meters) | Lower Decay Limit (sec) | Upper Decay Limit (sec) |
|---|---|---|
| 5.0 | 0.1 | 6.0 |
| 8.0 | 0.2 | 9.0 |
| 12 | 0.3 | 12 |
| 17 | 0.5 | 17 |
| 28 | 0.5 | 25 |
| 43 | 0.8 | 35 |
| 65 | 0.9 | 50 |
| 99 | 1.0 | 99 |

The company has planned for the future with this unit in the form of an empty port on the DN780's back panel. On future models, this port will house either an RS422 or RS232C interface, depending on user preference; the interface will also be available to current owners as a retrofit. K-T will provide the programing protocols for many of today's popular computers, which will give the user a great deal of remote flexibility. As of this writing, these interfaces are ready for market, and a MIDI interface is currently being worked through its last stages of development. Computer control will make the DN780 a powerful tool for film, stage, and multimedia, while MIDI will make it attractive to keyboard artists and composers.

## SUMMARY OF KLARK-TEKNIK DN780 DIGITAL REVERB SPECIFICATIONS

**Input**: one, electronically; 10 kohm balanced/10 kohm unbalanced.
**Output**: two, fully floating transformer balanced; 600 ohm minimum load impedance; source impedance less than 50 ohm; maximum level +21 dBm.
**Frequency Response**: 20 Hz to 12 kHz, +1/–2 dB.
**Distortion**: 0.03% at 1 kHz.
**Dynamic Range**: 85 dB, typical.
**Digital Processing**: 16-bit linear A/D and D/A converters; 32-bit arithemtic processor.
**Parameters**: zero to 990 mS pre-delay: decay time 0.1 to 99 seconds; room size five to 100 meters linear dimensions; HF/LF decay adjustable in 14 steps (relative to 1 kHz decay time); five variations of early reflection patterns, and level adjustable in 10 steps (0-max.)
**Power**: 100/120/220/240V 50/60 Hz; 40 VA.
**Weight**: 7.5 kg (net); 10 kg shipping.
**Dimensions** (W × D × H): 482mm (19 inch) by 310mm (12.25 inch) by 89mm (3.5 inch).
**Termination**: three-pin for input and outputs; three-pin CEE for power.
**Options**: transformer balanced input; PFR remote control.
**Price**: $5,500 including remote control.
**Factory and User-defined programs***: Factory-programmed hall, room, chamber, plate and effects settings in 28 dedicated memories. All settings can be called up by the keyboard, modified with the parameter controls, and stored for later use in one of the 50 user memories. Each memory stores all parameter settings and displays these when recalled.
*Hall (memories 1 thru 5):* early reflections of low density and diffusion give "depth and realism" augmented by slow attack and smooth decay.
*Plate (memories six thru 10):* High initial density and diffusion leading into smooth decay — a bright, clean attacking sound ideal for percussion and most contemporary music.
*Chamber (memories 11 thru 15):* The uneven, moderately dense early reflections produce a "bright, lively" sound midway between hall and plate.
*Room (memories 16 thru 20):* Short, high-density early reflections with medium to fast attack and high diffusion produce "authentic room simulation" for drama, film dubbing and ambience applications.
**Special Effects**: *Straight 0 to 2 second delay; Multi-tap Echo; "Infinite Room,"* an electronic "zero-absorption" space in which sound is continually reflected; *ADT* with extra taps for creating choir effects; and *Sound-on-Sound* with control of loop length and erasure; plus *Alive, Non-linear Decay* and *Reverse Decay*.
*According to the manufacturer, purchasers of the DN780 are entitled to receive new programs in the form of plug-in updates on an EPROM. Hinged circuit boards and a zero insertion force socket are said to simplify the installation of new programs when they are received.

**Manufacturer: Klark-Teknik Electronics, Inc., 262a Eastern Parkway, Farmingdale, NY 11735. (516) 249-3660**

### Studio Experience

I spent a fair amount of studio time with the DN780, finding it to be an asset both for cutting tracks and mixing. Almost every factory-loaded program had merits for some type of material, which made it quite simple to throw together some fast reverb for a musician's headphone feed or for control-room playbacks. It also made the mixing process run quickly, since parameter adjustment was simple, with many variations sounding good.

I ran the listening test with rock material consisting of drums, bass, guitars and keyboards containing both staccato and sustained lines, and male and female vocalists. Since I found most of the factory presets to be very good, my approach was a bit unorthodox. After playing with the program and doing minor adjustments, I changed the Patterns, thus changing the entire nature of the beast. I was surprised to find that in many, but not all, cases Pattern changes yielded a program that was still desirable. I would have to attribute this to Klark-Teknik's stated design philosophy of having no exclusive algorithm restricted to a particular program — all parameters are interactive on the reverberation programs.

The **Plate** programs were probably the most impressive ones in the unit, and they gave a a good true plate sound and had a number of good variations.

*Plate 1:* Very short and dense, this program added presence to vocals, and worked well with guitar and keyboards; even kick worked here.

*Plate 2:* A bit longer with more low-end, this was nice on vocals and sustained keyboard patches. Changing to Pattern #3 or #4 gave me some good guitar reverb.

*Plate 3:* Noticeably longer and not as bright as the above plates, low-end material such as kick, and richer tones sounded good here. Vocals and guitar worked very nicely.

*Plate 4:* A longer RT60 of two seconds gave the feeling of space. The setting was nice and bright for vocals and keyboards sounds extremely good.

*Plate 5:* Longest RT60 of 3.6 of all plates, this program was rich with lows and mids. All drums sounded very effective, and a very large space was created for vocals and guitar. Keyboard sustains had a big sound.

*Hall 1:* A small hall, this gave presence and a feeling of doubling to keyboards. Lower harmonics were brought out, and gave a good depth to guitar and vocals.

*Hall 2:* Although not thrilling, snare and sustained keyboards worked well. By dropping the pattern to #1, as I increased the pattern number I could hear the "walls" move away from me.

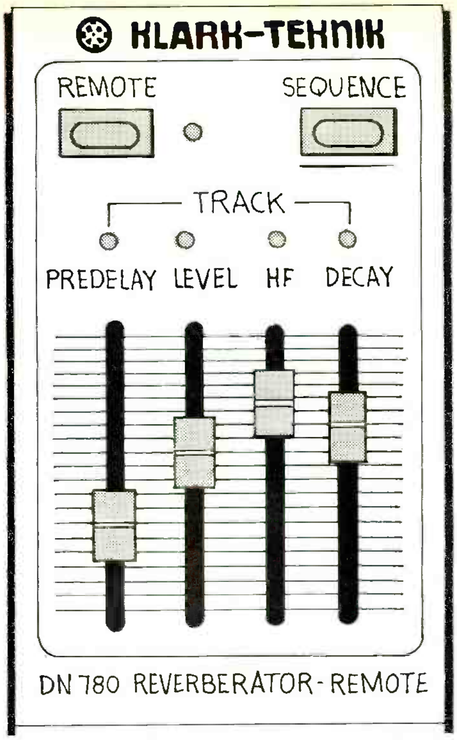
*Hall 3:* This was generally a very good program with some big keyboard spaces.

*Hall 4:* One of the smoothest halls, vocals sounded nice with a very long decay. Keyboards sounded very bright and tight when using lower Patterns.

*Hall 5:* A huge room with extreme decay, it is sensitive to percussive sounds, wind noise, harmonic distortion, fret noise, keyboard FM noise and vocal pops. Experimentation is needed for special effects.

The really big halls such as #4 and #5 demonstrate problems that I have run across in several digital reverbs. Sensitivity to percussive and high-frequency information can sizzle, and sound quite harsh. In these cases, I generally found that by placing a low-pass filter on the echo send to the DN780, rolling off some of the highs and then increasing the HF parameter on the front panel, I was able to process this material without losing too much top-end.

The **Chambers** were really quite




REMOTE-CONTROL PANEL LAYOUT

good, and ranked after the Plates as my favorites. They were nice and bright and simulated the chamber sound realistically.

*Chamber 1:* This chamber is very small and dead, producing an interesting effect on small, tight vocals, which could be utilized for emphasis.

*Chamber 2:* Bright and very effective, this setting was really nice on all material. Lowering the pattern provided a great kick sound.

*Chamber 3:* A medium-size chamber that did well on snare, vocals, and guitar; keyboards were big and rich.

*Chamber 4:* This is a larger chamber that worked well with vocals and guitar. By lowering the Pattern, I produced some real good spaces for keyboards and percussive material.

*Chamber 5:* Really big, yet smooth. Great for background vocals, especially "Ooos" and "Ahhs," sustained key lines and strings.

The **Rooms** were effective at simulating boundaried spaces, and could be applied to film soundtracks as well as music sessions. I did a lot of Pattern changing within these programs, finding some nice overtones and harmonics could be added to keyboards and guitar in the smaller rooms.

*Room 1:* This is definitely a small room with hard walls easily discernable.

*Room 2:* Still fairly small and fast, vocals and keyboards did best.

*Room 3:* A mid-sized room that responded very well to kick when the HF was dropped down a bit. Increas-

## KLARK-TEKNIK DN780 REVERB

ing the Pattern added a lot of depth to background vocals and guitar.

*Room 4:* The same size as Room #3, but with a longer decay and less early reflection level, this was a strong choice for lead vocals. Keyboards and strings shown through here too.

*Room 5:* The biggest Room, yet still manageable, with great kick and snare possibilities, as well as staccato keys.

There are three currently popular reverb programs also stored in the DN780: **Alive, Gated Decay,** and **Reverse Decay.** These special-effects reverb programs are not governed by the Room Size or Pattern parameters. Instead, the Pattern button functions to change the decay envelope as Linear decay (medium), Gated decay (fast) and Reverse decay. The Level button then controls the reflection density of the effect.

*Alive* is a non-linear program that's a dead ringer for that well-known British unit. It is very versatile with all its parameter flexibility, and takes lots of level from drums. The three densities create a tight, non-linear pattern, a longer pattern, and a reverse-envelope pattern.

*Gated Reverb* was adequate, but not

**KLARK-TEKNIK DN780 FACTORY LOADED REVERB PARAMETERS SERIES ENABLED BY V1.5 SOFTWARE**

| Memory Number | Program Title | Pre-Delay (mS) | Pattern | Level | Decay (sec) | LF | HF | Room Size (meters) |
|---|---|---|---|---|---|---|---|---|
| 01 | Hall 1 | 37 | 4 | 9 | 0.7 | +1 | +1 | 28 |
| 02 | Hall 2 | 47 | 4 | 5 | 1.4 | +1 | 0 | 43 |
| 03 | Hall 3 | 116 | 5 | 6 | 2.8 | +1 | 0 | 43 |
| 04 | Hall 4 | 76 | 4 | 4 | 3.6 | +1 | 0 | 65 |
| 05 | Hall 5 | 172 | 5 | 6 | 6.0 | +1 | −2 | 99 |
| 06 | Plate 1 | 00 | 1 | 6 | 0.4 | −3 | +4 | 8.0 |
| 07 | Plate 2 | 00 | 1 | 4 | 1.0 | 0 | +4 | 17 |
| 08 | Plate 3 | 00 | 1 | 1 | 1.4 | 0 | +1 | 43 |
| 09 | Plate 4 | 00 | 1 | 0 | 2.0 | +1 | +1 | 43 |
| 10 | Plate 5 | 00 | 1 | 0 | 3.6 | +3 | 0 | 43 |
| 11 | Chamber 1 | 25 | 3 | 9 | 0.4 | 0 | +1 | 12 |
| 12 | Chamber 2 | 32 | 3 | 9 | 1.0 | 0 | +3 | 28 |
| 13 | Chamber 3 | 55 | 3 | 9 | 2.4 | 0 | +1 | 43 |
| 14 | Chamber 4 | 65 | 3 | 9 | 3.2 | 0 | +1 | 65 |
| 15 | Chamber 5 | 65 | 3 | 7 | 8.0 | 0 | −1 | 99 |
| 16 | Room 1 | 00 | 2 | 9 | 0.1 | 0 | +4 | 5.0 |
| 17 | Room 2 | 32 | 2 | 9 | 0.3 | 0 | +4 | 8.0 |
| 18 | Room 3 | 35 | 2 | 9 | 0.8 | 0 | +2 | 17 |
| 19 | Room 4 | 53 | 2 | 6 | 1.2 | 0 | +2 | 17 |
| 20 | Room 5 | 45 | 2 | 4 | 2.8 | 0 | +1 | 28 |





For additional information circle #196

impressive. I would prefer to gate a really good, applicable reverb program, but yet to find a digital reverb unit that has a gated program as effective as outboard gates.

*Reverse Decay* is a good program, and was fun to use for drums and vocals. The lowest levels for reflection density were the smoothest, most effective settings.

Five **Special Effects** programs top off the factory-stored programs:

*Delay* is a simple delay line variable up to two seconds; it is clean with little high-frequency loss.

*ADT* stands for automatic double tracking, yet much more is possible here. Delay is variable up to 127 mS, and the number of voices from two to eight. The program is capable of nice, widespread multivoice delay effects.

*Multi-Tap Echo* didn't impress me, and I found it not very easy to find an effective echo pattern. It could be effective, however, as a cluster of echoes going to a plate or chamber.

*Sound On Sound* allows you to capture up to two seconds of program and then record over it. This was nice just for a one effect program, but limited in usage since no real playback control is provided. Adding sounds on top of each other has a disadvantage of not being triggerable, and so building sounds is difficult.

*Infinite Room* is a very nice effect. Although it lacks the "air" of its German counterpart, it still has body and warmth. It creates reverb in a theoretical room with no loss, and one can load many sounds into the program to create something unique.

After my tests, I gave the unit to Rick Sanchez to see if his experiences would match my analysis. Sanchez, an independent engineer, was a long time Record Plant employee who worked on a wide variety of projects with such artists as Rick James, The Wispers and Survivor. He was most impressed with the Plate settings, and felt that the factory presets were better than most other reverbs he has worked with. "It was very usable," he enthused. "The decay time and frequency were what I expect from a plate, and very little parameter manipulation was needed."

Sanchez also liked the special effects reverb programs, and thought the short rooms to be impressive. He stated that the unit was very easy to operate, which made it a desirable tool. Klark-Teknik has certainly done its homework on programs, and come out with some very nice sounds.

With the settings offered by the factory, combined with the 50 user-storage locations, an engineer should be able to make the DN780 a very creative package. I have also been informed that, by the end of the year, there will be some new factory programs and updates available.

Maintenance engineers will also like the DN780 for its simplicity of internal design and diagnostic routines. When the cover is lifted one is impressed with the clean, uncluttered layout. The circuit boards are of a two-layer design, and the top PC board attaches with three thumbscrews, and opens up on a hinge like a car hood. If necessary, repair can be effected quickly.

Upon power-up, the DN780 runs a quick self diagnostic, and then resets to the program that was in memory at the time of power-down. If there is a problem, 11 separate tests may be run to pinpoint the specific problem. Even remote-slider noise can be tested to check if it is within tolerances. It is extremely important that a manufacturer supports complex digital devices for its owners, as Klark-Teknik has done so by providing as much user maintenance information as possible.

All things considered, the DN780 turned out to be fun to review, and a real surprise considering its moderate cost. It is a very well thought out piece of equipment, and I would not hesitate recommending that you give this a good listen when shopping for reverb.

I would like to thank Studio D in Sausalito, CA, for providing the time necessary for this in-use operational assessmet.

# URSA MAJOR MSP-126 MULTI-TAP STEREO PROCESSOR


*Reviewed by Bob Hodas*

The MSP-126 Multi-Tap Stereo Processor is designed for use by both the broadcast industry and the recording studio. As its name implies, the device is a stereo digital processor that utilizes a 12-tap algorithm or controlling "program" to control the various output signals generated from a mono input. Besides stereo simulation, the MSP-126 also features several special effects modes that take it beyond a normal DDL operation. Processing is achieved with two groups of six taps, each group being assigned to a separate output. There are eight separate programs or Modes, each with two control Parameters incorporating a number of variations. The processor utilizes a 15-bit PCM (pulse-code modulation) analog-to-digital conversion process; sampling frequency is 44.1 kHz.

Rear-panel inputs are via two, XLR-style connectors, and may be run with stereo or mono input mode that is controlled via a switch. The provision of stereo inputs allows (particularly in broadcast situations) a stereo signal to pass through the unit unaltered in bypass mode, with the option of switching out of bypass to process an incoming mono signal. There are two XLR-style output connectors, and a quarter-inch phone jack for a bypass footswitch. The unit can be operated in either balanced or unbalanced mode.

The front-panel layout is very simple and straightforward, and includes input and output level pots, a five-segment LED input-level indicator, a Bypass switch with LED indicator, and a power on/off switch. A 16-character display window defines the Mode, Parameter #1, and Parameter #2 with red alphanumeric LEDs; the latter are less than quarter-inch high, but very bright and easy to read even in strong light. The three knobs for Mode and Parameters are fully rotating digital controls, and can sweep through the settings in either direction, which facilitates a speedy setup when looking for a specific setting. The Mode switch enables the user to select from among the eight different programs stored in the unit. The Parameter #1 switch can select among 16 delay variations in each Mode, while the Parameter #2 switch selects from the 16 gain variations available in each Mode. As you can see, there are a finite number of variations possible with the MSP-126 — a total of $8 \times 16 \times 16 = 2{,}048$ combinations of program, delay and gain — yet this does not seem to be a restriction.

### Operating Modes

Two of the operating Modes are designed to process mono signals into stereo, and are aimed mainly at broadcast applications. The stereo synthesis capability will come in very handy as the U.S. turns increasingly to stereo television sound, and there develops the need to process older, mono programs for a more "spacious sounding" broadcast.

The other six Modes are quite specifically recording- or mixing-oriented programs, although the design of the MSP-126 processor makes its application different from most other outboard gear. Instructions from the manual recommend that you do not connect the unit in an effects send/return configuration, because undesirable comb filtering can result in certain Modes if the output is mixed with the direct signal. Since the input signal is converted into a digital format and written to memory, the direct signal is delayed on the order of 50 to 70 microseconds. In some cases this was not a problem, and I simply patched the MSP-126 from the machine out to channel input or in-line via an insert patch point. At other times though, I found myself desiring more direct signal, but was restricted and could not achieve the proper direct-to-effect mix ratio. When this was the case, I tried the unit in a side-chain configuration, and found that in a few cases the MSP-126 operated just fine in a normal send/return configuration. It is a matter of taking some time to find which situations work best with which particular configuration.

Let's now examine the different Modes in greater detail:

- **MSP-11 (Multi Tap Stereo Processing)** creates a stereo output from mono input sources with a flat response. Parameter #1 represents "Depth" (D5-D100) and Parameter #2 "Width" (W0-W100). I found a combinaion of D45 and W100 to be a very desirable setting for processing mono into stereo. Lower depth values were generally more convincing, with a noticeable phase and image shifting occurring at settings above D60. The Width parameter is effective in controlling stereo spread, although I usually like the widest spread possible. In the recording studio this could be a good tool where mono compatability is an important consideration.
- **CSP (Comb Filter Stereo Processing)** creates stereo from mono with complimentary comb filters. Parameter #1 adjusts the frequency of the first comb peak and null (opposite channels), while Parameter #2 sets the depth of the comb. By setting P1 at either 200 Hz or 4.8 kHz, I obtained a well balanced stereo spread but, at other settings, the image pulled to the left or right, varying with the extremes of frequency — often I got the feeling that frequency bands were split between the speakers. Although this effect could be interesting for guitar or synthesizer effects, I'm not impressed with its stereo process.
- **ROOM** provides early reflection patterns of rooms. Parameter #1 adjusts the longest reflection time from 5 to 360 milliseconds, while Parameter #2 adjusts the ration of Dry to Wet signal at the outputs in percentage (0 to 100% reflections). I created some really fine, non-linear snare reverb

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For additional information circle #198

## SUMMARY OF URSA MAJOR MSP-126 STEREO PROCESSOR SPECIFICATIONS

**Bandwidth**: 20 kHz, 44.1 kHz sampling rate.
**Noise**: +80dB dynamic range; 15-bit PCM conversion.
**Size**: Rack mount, 19 by 1.75 by 11 inches, (W×H×D); weight, approximately 12 lbs.
**Inputs**: Electronically balanced (differential amplifier); pin #3 high, pin #2 low and pin #1 ground. *Stereo input mode*: input impedance of pin 3 is 11 kohms and pin 2 is 21 kohms; in mono input mode, input impedance of pin 3 is 5.5 kohms, pin 2 is 10.5 kohms; recommended source impedance is 600 ohms or less. Maximum source voltage before input stage overload is +17 dBv (7 VRMS); minimum input for operation of "zero LED" (input level fully clockwise, input frequency 100 Hz; *Mono input mode*: (left input only driven), is -10 dBv (0.316 VRMS); in stereo input mode, with both inputs driven in phase, sensitivity rises to -16 dBv (0.159 VRMS); connectors are XLR-3 female.
**Outputs**: In stereo active differential circuit: pin #3 is high (100-ohm source resistance); pin #2 is low (100-ohm source resistance; minimum recommended load impedance is 600 ohms; output stage maximum level is +17 dBv feeding an unbalanced load, and +23 dBv feeding a balanced load; connectors are XLR-3 male.
**Bypass Function:** *Stereo input mode:* bypass "on" feeds the left input directly to the left output, and the right input directly to the right output; *Mono input mode:* bypass "on" feeds the left input through to both left and right outputs.
**Input Mode**: *Stereo input mode:* the left and right input signals are summed and fed to the processor; *Mono input mode:* the left input only is sent to the processor.
**Parameters**: *Parameter #1:* 16 delay variations of each mode; *Parameter #2:* 16 gain or amplitude variations; a 16 character, alphanumeric display shows the selected mode for both parameter settings.
**Mode Set MSP-11**: The first "Mode" set, or program set consists of eight modes —*MSP11 (Multi-Tap Stereo Processing):* User adjustable time delay patterns to create a full, stereo image from a mono source with flat response and complete mono compatibility (left, right, and L+R); usable with a wide range of program source and adjustable from mono to full width, and for "depth." *CSP (Comb Stereo Processing):* Creates stereo using complementary comb filters, with L+R compatibility; adjustable for comb frequency and stereo "width." *ROOM (Room Early Reflections):* Generates the early reflections of a room or concert hall, with adjustable "size" and dry/wet mix. *DELAY (Delay Clusters):* Provides a cluster of delay repeats, with adjustable pre-delay and dry/wet mix; realistic slap and echo effects. *PAN (Pan Pot Using Delays):* A function that places a source "anywhere" in the stereo field using time delay patterns — equal signal energy in both channels at all times; adjustable for position and image width. *DDL (Digital Delay Line):* Used as a simple DDL with adjustable time delay from 100 microseconds to 360 milliseconds. *RPTS (Repeats):* Generates from two to 10 equally-spaced repeats, alternating between channels, with adjustable overall time delay, and rising or falling gain. *SCALE (Filter in Musical Scale Steps):* Provides a stereo comb filter with "teeth" that are at precise musical intervals; adjustable up a chromatic scale from unison to an octave plus a minor third.
**Price**: $2,000 (suggested list).

**Manufacturer: URSA MAJOR, Inc.**
**P.O Box 28, Boston, MA 02258. (617) 924-7697**



with P1 at 200 milliseconds and P2 set to 30 to 60%. Setting P1 between 300 and 360 milliseconds and P2 at 100% was also really fun. The snare was side-chained, and the MSP-126 output fed to a live chamber then mixed with the direct snare. Although we did not have a plate to use during this assessment, I would be interested in seeing how much such a device could be improved with Room at the front-end.

• **DLAY (Delay Cluster)** provides a cluster of delays after the initial delay. Parameter #1 adjusts the time to the beginning of the delay cluster (from 20 to 320 milliseconds), while Parameter #2 adjusts the loudness of the cluster in relation to the dry signal. Some interesting vocal doubling effects were achieved with this program. P1 was set to 40 milliseonds, and P2 to between 0 and 80% and then sent into the chamber. I didn't feel this program was as effective with percussive material, and not as smooth as Room for creating pre-delays to reverb.

• **PAN** places the input signal in a left-to-right position using delays. Parameter #1 adjusts the Left-Center-Right positioning, while Parameter #2 adjusts the apparent width of the image. Pan is an effect that one would find useful if you were concerned with mono compatability, and hard panning created to much out-of-phase information.

• **DDL (Digital Delay Line)** is a simple delay line that appears Left, Right, or Left+Right. Parameter #1 is a delay-time multiplier from 1 to 16, while Parameter #2 adjusts the delay time and position from 1 to 20 milliseconds Left or Right, and 0.1 to 20 milliseconds L+R. This Program is simply



For additional information circle #199

a DDL with position with a maximum delay time of 160 milliseconds. It worked fine, but I needed to mix in some direct signal for a good rock echo effect when working in the L+R position.

• **RPTS (Repeats)** gives stereo delay repetitions: Parameter #1 sets the length of the last repeat from 10 to 360 milliseconds, and Parameter #2 sets the number of total repeats from 2 to 10, as well as whether the repeats increase or decrease in volume or remain equal. This was a nice program capable of a variety of effects. By setting P1 at 320 milliseconds and P2 at 10 taps increasing, I achieved some good backward echo-type effects. A strong hard rock vocal was also easy to dial in with this program. I also obtained a unique effect by sending the output into a chamber using a snare as the signal. At different delays the snare excited different harmonics in the chamber, and gave a tuned ringing effect.

• **SCALE** is a stereo comb filter with musical intervals L-to-R. Parameter #1 sets the musical interval between the fixed filter (A440) in the left channel, and the variable filter in the right channel (from A, Unison up through a chromatic scale to the third above A880). Parameter #2 sets the filter intensity from peaks at all harmonics (eight settings) to peaks at only odd harmonics (seven settings). This is specifically a special effects program and some experimentation is required.

Operating the MSP-126 was a pretty simple matter, which says a lot for fast effects in the studio. One annoyance was that I could not make adjustments in one Mode without changing Parameters in another. This prevented me from comparing the effect of two programs easily, when deciding whether to use RPTS or DLAY in front of the chamber. Some programs seemed more applicable than others, but that is a highly subjective matter. I personally found three programs that were impressive.

A large, competitive market exists for special-effects devices, but Ursa Major has centered on the mono-compatability aspect of this market. The MSP-126 equipped itself well in terms of its ability to convert a mono signal into a mono-compatible stereo output with interesting spatial dimensionality — an essential requirement for units that are intended for TV and radio broadcast applications — in addition to providing an adequate range of delay-related special effects. ■■■

**I want to thank San Francisco Sound Recorders for donating the time needed to evaluate the MSP-126 for this in-use assessment.**

# New Products

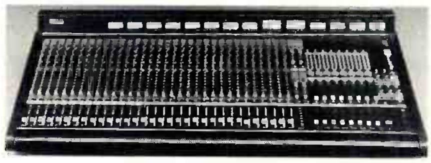
## YAMAHA LAUNCHES PM3000 PRODUCTION AND LIVE-PERFORMANCE CONSOLE

Unveiled at the recent AES Convention in New York, the Yamaha PM3000 console is available in three configurations of 24-, 32-, or 40-input channels, and incorporates improved input circuitry with a five-position attenuation pad switch and gain control; eight VCA groups; eight group mixing busses and eight auxiliary mixing busses; discrete stereo bus; an 11-by-8 mix matrix configuration; eight master mute groups; and extensive cue and solo capabilities.

All of the conventional auxiliary and group busses may be operated independently, resulting in a total of 18 discrete busses, including the stereo bus. Additionally, a total of 26 mixes are effectively available by re-setting the mix-matrix internal preset switches. The mix matrix, pioneered by Yamaha, permits 11 possible sources to be mixed together eight different ways on eight different modules.

New in the PM3000 is a VCA grouping system distinct from the audio grouping, providing a capability that goes beyond the group busses, and even permitting outboard control of group level controls. Eight VCA switches next to each channel fader enable that channel to be controlled by one or more VCA master faders. With multiple input channels assigned to a given VCA master, all of those channels' post-fader output levels can be raised or lowered by the VCA master fader.

Furthermore, certain applications may call for using the eight VCA groups as though they were discrete busses, giving the PM3000 the functional equivalent of 26 separately-controlled groups. This feature can be particularly valuable, for example, in a situation that calls for live-performance and recording simultaneously; using the VCA groups as discrete busses will effectively double the console's capacity.



Inputs are differentially balanced, equipped with a five-position attenuation pad and a continuously variable gain trim control. Where extra grounding isolation is required, optional onboard IT3000 input transformers are available. Each input channel includes a four-band parametric equalizer with EQ in/out switch, permitting instantaneous bypass of the entire circuit. The 12 dB per octave highpass filter on each input channel has its own in/out switch, and its –3 dB cutoff frequency has a wide sweep (from 20 Hz to 400 Hz).

Also new with the PM3000 is a master mute function: each input channel has eight mute assign switches that permit the channel's on/off function to be remotely controlled by the eight master mute switches. Cue and solo capability includes cue/solo switches on every input channel, and a cue switch on every master auxiliary send, group outputs, stereo master outputs and the auxiliary returns.

Four auxiliary returns are available for effects, or to be used as mono or stereo line inputs to the console. Each return channel is stereo, and comes equipped with its own two-band, shelving, sweep-type EQ, balance control, cue and on/off functions, and may be assigned to every bus on the console. Extensive metering is provided with 14 VU meters, each with a peak LED, that can be switched to monitor 35 different points. Numerous LEDs indicate status, display clip levels, and are used for illuminating all switches.

**YAMAHA INTERNATIONAL CORPORATION**

**For additional information circle #202**

## NIKKO NCD-600 DIGITAL COMPACT DISC CHANGER

Billed as "the most sophisticated digital disc player on the market today," the NCD-600 can be programmed to randomly access any sequence of selections on up to 60 simultaneously stored Compact Discs, permitting 45 hours of continuous-play programming in any disc order, or up to five selections in any selection order. The unit can also be fiber-optic linked to other NCD-600s, expanding program access to as many as 240 discs.

The system's three-beam laser pickup interfaces with an RS232 serial interface and eight-pin DIN RGB input on the side panel. In addition, the NCD-600 can also be interfaced with the new CD ROM.

Featuring key control and tempo maintenance control between selections, the NCD-600 is said to be versatile in a variety of applications from commercial sound-effect libraries to professional studio/reinforcement.

Having established an "Advisory Board of Audio Engineers," the company recently placed a few units in the field for testing. Film sound/designer/musician/composer, Frank

Serafine, says that the CD Changer has substantially reduced research time. "I was recently selected to work on Disney's upcoming *Navigator*. One of the contributing factors in my favor is my NCD-600's ability to cut down the time it takes to research music, sounds, and effects," he offers. Serafine, who claims to have digitally recorded over 2,000 of his own sound effects in past years, is now converting them to Compact Disc for storage in a bank of NCD-600s. "Up until now, whenever I needed a sound or was looking for an effect, I had to thumb through a stack of indexes, pull the disc, and load up to listen. Now with the NCD-600, I can touch-in index numbers as I scan my files, listening to one disc while loading up others. This new way of researching music has made my life a lot simpler, allowing me to focus my energies and time on creating."

**NIKKO AUDIO**

**For additional information circle #203**

### J.L. COOPER MODEL 16/20 MIDI SWITCH BOX

Containing 16 MIDI inputs and 20 MIDI outputs, the programmable MSB 16/20 is described as being invaluable for the larger MIDI setups, and particularly Yamaha QX-1 and TX-816 owners.


The 16/20 can hold 64 "patches" (configurations) in its battery-backed memory; changing configurations is done via front-panel buttons, or by MIDI program-change commands from a master synth or sequencer.

Via MIDI, the unit can load its configuration to a master computer, thus having the computer play in words the configuration; for example "QX-1 Track 2 to Prophet V."

Suggested retail price of the MSB 16/20 is $1,395.

**J.L. COOPER ELECTRONICS**

**For additional information circle #204**

### AUDIO KINETICS PACER TIMECODE CHASE SYNCHRONIZER

The Pacer is a single-rack space unit that works with all international timecodes, and features an international generator capable of jamsync operation locked to house sync/external video source. The twin dynamic readers are capable of reading code from 1/20 to 80 times play speed in both directions.

The hard-lock system with chase capability offers slave-machine control with an optional control pad or via the RS 232C/422 computer port. Also featured is Automatic Calibration with battery back-up, and Automatic Offset calculation, which can be manually trimmed by frame and subframe. The system is designed for a wide range of applications, from locking up two audio transports, to adding audio to a control track editing system or a ¾- or one-inch video production suites.

No prices are currently available, although the market price is expected to be $3,000 or less.

**AUDIO-KINETICS, INC.**

**For additional information circle #205**

### LEXICON PCM-70 DIGITAL EFFECTS PROCESSOR WITH DYNAMIC CONTROL

Software packages, in the form of plug-in chips, can be changed as new programs become available. The first software package for the PCM-70 allows the machine to function as a digital effects processor that will be able, for example, to produce hundreds of sounds in time delay or digital reverb. Lexicon's Dynamic MIDI feature is fully compatible with other MIDI equipment, such as sequencers and computers, and enables full remote control through a MIDI keyboard or other MIDI control devices.

With Dynamic MIDI, as many as 10 effects or reverb parameters can be assigned to any MIDI controller or keyboard, and varied dynamically via key velocity, pressure or aftertouch.

The newly-developed digital effects and reverb programs for the PCM-70 are derived from algorithms found in more expensive Lexicon digital effects processors. More than 80 different parameter types, such as delay times, beats per minute, feedback, wet-dry mix, high- and low-pass filters, room size, etc., are contained in the programs.

Thirty separate programs covering chorus and echo, resonant chords and multi-band delays are incorporated in the initial software package. For reverb the PCM-70 offers new versions of such well-known Lexicon reverb programs as Rich Chamber, Rich Plate, Concert Hall, and Infinite Reverb. In addition to the more than 40 effects and reverb programs, 50 programs can be created, named and stored in the user registers — providing a total of more than 90 instantly-available programs.

Factory programs comprise seven families: chorus and echo, multiband delays, resonant chords, concert hall, rich chamber, infinite reverb and rich plate. The chorus and echo program family offers six separate chorusing voices with individual stereo-panable locations, and more than 45 parameters.

The multiband delays provide six separate panable voices, each with a separate HF and LF, and more than 50 parameters.

The resonant chord programs turn the PCM-70 into a six-voice resonator, with each voice tunable over a seven-octave range. Via MIDI, the notes can be played by assigning them to a keyboard.

Front panel controls consist of dedicated keys and a "soft" knob which can be assigned to as many as 10 separate control parameters simultaneously to create a "Grandmaster Control." Each patch has a separate scaling factor, positive or negtive, that enables the soft knob to be fine-tuned for one parameter while simultaneously coarse-tuning another.

Patches can be assigned internally, or to external devices via MIDI protocol. MIDI controller data can be recorded on a sequencer, for editing and subsequent playback into the PCM-70. In addition, the unit incorporates a corresponding register table that allows all 128 MIDI-specified presets to be utilized in recalling the PCM-70's programs and user registers.

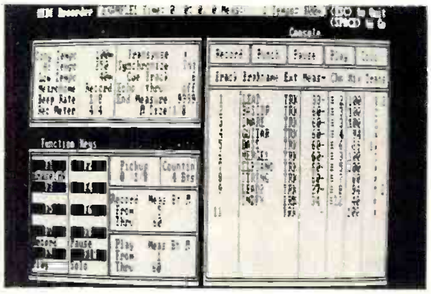
**LEXICON, INC.**

**For additional information circle #222**

## MIDI ENSEMBLE SOFTWARE FROM SIGHT & SOUND MUSIC

Designed to run on IBM-PC and compatibles, MIDI Ensemble consists of three main program modules: Recorder, Event Editor, and Phrase Editor.

The Recorder module lets the user record and overdub tracks like a multitrack, and includes automated punch-in and -out, autolocate, programmable metronome, track transpose, elapsed time measurement (real-time or frames), solo/mute tracks, pause, cue, etc. Interfaces are provided to external controllers, including tape sync, MIDI Clock and Song Position pointer. A total of 255 tracks are available for recording.



The Event Editor is specifically designed for editing music tracks in microscopic detail, and can be used to insert, remove, or change notes in a track. A special screen display is said to display simplify the precise editing of pitch, start time, duration, and key-strike velocity of any note, and the user can also insert, delete, or adjust the timing of program changes and phrase markers.

In the Phrase Editor, the beginning and end points of music segments can be specified, and then moved, copied, deleted, combined and modified in various ways. Segment (phrases) can be any length, from an entire track to just a part of a measure. One phrase can be replaced with another, phrases merged together and autocorrected.

Suggested user price of MIDI Ensemble is $425.

**SIGHT & SOUND MUSIC SOFTWARE, INC.**

**For additional information circle #223**

## NEW MPX-2000 PRODUCTION AND BROADCAST CONSOLE UNVEILED BY SONY

Designed for both on-air and post-production use, the new console is said to spearhead an increased commitment by Sony Professional Audio Division to supply products that specifically address the needs of broadcasters and television post-production facilities. According to George Currie, the Division's VP and general manager, "With Stereo TV a reality, and greater emphasis now being placed on audio quality throughout the broadcast chain, we're even more committed to bringing state-of-the-art equipment to the broadcast industry."

Special features are described as enhancing the console's utility in the control room are joined with the extensive inputs and outputs, grouping functions, etc., needed for creative ease in stereo audio-for-video post production. Among the MXP-2000's features for on-air use is optional user-assignable dynamics processor module that provides compressor limiting or expansion of any input. For post-production, the console has facilities for a planne video editor interface. When released, the optional interface will allow the console to work with certain editors or switchers to control crossfades and mutes to within frame accuracy.

The unit also offers switchable dual mike/-line inputs, five different kinds of solo, selectable audio groupin or VCA grouping, cough switching, three-band EQ sweepable from 300 Hz to 3 kHz, direct outputs for direct multitrack sends, group outputs, and on-air/rehearsal logic switching. Stereo capabilities include two independent stereo outputs, and four stereo external monitor inputs. Stereo line inputs are planned. The standard 20-module frame allows total interchangeability; any type of module will function in any slot, thus allowing any physical arrangement of console functions.

Group and input/output modules may also be mixed according to the user's requirements. Up to four group modules may be installed for control of 12 input channels or, if groups are not used, a maximum of 16 input channels is possible.

Circuit-card jumpers enable switching between transformer or electronically balanced inputs.

**SONY CORPORATION OF AMERICA**

**For additional information circle #224**

## STUDER ANNOUNCES A820 HALF-INCH TWO-TRACK AND TIMECODE VERSIONS

The half-inch two-track A820 is designed primarily for extremely high quality music mastering applications. The added tape width of the A820-2-½ is said to provide significant improvements in both signal-to-noise ratio and tape saturation characteristics.

The A820-TC features the center-track SMPTE timecode system first introduced on the A810. Two separate heads are used to record and reproduce the timecode, thus keeping code/audio crosstalk at better than a quoted -90 dB. A microprocessor-controlled delay line compensates for the offset created between code and audio heads, thereby maintaining exact coincidence of code and audio at all speeds, including vari-speed modes.

The basic A820 transport handles 14-inch reels, offers four speeds (3¾, 7½, 15 and 30 ips), and microprocessor-based functions, thereby permitting software control of virtually all operating parameters. Approximately 40 different user programmable functions may be assigned to various keys. All audio parameters are digitally set and stored in non-volatile memory.

Professional user-net price of the A820-2-½ is $11,000; the A820-TC is $11,500.

**STUDER REVOX AMERICA, INC.**

**For additional information circle #225**

## EDITRON INTRODUCES A RANGE TIMECODE SYNCHRONIZING AND EDITING SYSTEMS TO U.S. MARKET

Consisting of an inter-related family of products for sound, film and television industries, the three synchronizers in the company's line range from a simple three-machine operation, through to a more complex 15+ machine operation. Each of these can form part of a large synchronizer network under the control of the two large intelligent control systems: the Model 200 and 500.

The smallest of the family, the Model 100 system, offers extensive front-panel controls, and is said to provide a highly sophisticated, entry-level synchronizer/chase system.

**EDITRON AUSTRALIA PTY, LTD.**

**For additional information circle #226**

# New Products

### MIDI INTERFACE FOR LEXICON 224XL REVERB FROM ELECTRON FARM

The new MIDI/224 Interface provides both program change *and* continuous control of all LARC parameters from a MIDI synthesizer or computer-based sequencer.

The following functions can be achieved:

- Change 224XL programs from a MIDI sequencer or synthesizer program switches.
- Control LARC faders from MIDI control source with complete programmability — assign any MIDI controller to any combintaion of control parameters on the current 224XL page.
- By-pass mode returns full control to the LARC.
- Last or average key number, velocity and pressure.
- Receive on any MIDI channel.

The MIDI/224V combines all of the features of the MIDI/224 with eight control voltage or resistance outputs to retrofit existing processors and synthesizers to accept MIDI control.

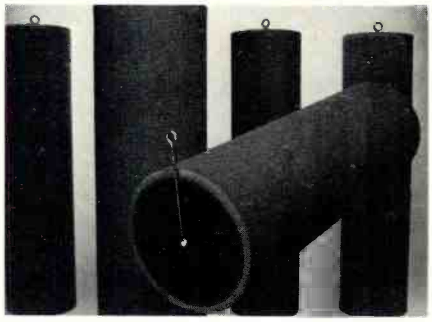
Either unit is installed between the 224 mainframe and the LARC controller, and comprises a microprocessor that translated the MIDI data to LARC format and, on the MIDI/224V, eight DACS to provide MIDI-to-CV translations (to enable MIDI control of other studio equipment). In bypass mode, all controls are from the LARC; when the unit is in operation, Program Selection and the LARC fader data are derived from the MIDI input, while Page selection and display remain at the LARC.

**ELECTRON FARM**

**For additional information circle #228**

### ASC UNVEILS "TUBE TRAP" LOW-END ACOUSTIC CONTROLLER

The Tube Trap is a modular line of small, efficient, lightweight, portable and patented bass traps. THe three-foot tubes can be used alone, stacked into columns, installed with wall or ceiling hangers, or used as a studio gobo system.



Comprising a pressure-zone rap, the units are said to be best utilized when backloaded by a wall or corner; tri-corner installations in rectangular rooms are described as effectively dampening major room resonant modes.

Two models are available: M-90 voice-range trap (nine inches diameter to provide 10 sabines per three-foot section absorption; and the M-45 trap that ranges an octave below ti 45 Hz (11 inches in diameter providing 15 sabines per section).

Tube Trap installations are said to clean up transient distortions due to room acoustics; low-end phase shifting in the attack of a tone burst is reduced by trapping the first, strong reflection. Room resonance modes that color the tone burst decay, muddy the sound and cause a low end build up are also controllable. The result described as is a "faster and more detailed" low-end acoustic room.

All units have a Midrange Crossover Diffuser Panel section that keeps the sound field balanced even when listening in the close-field of the absorption unit; rotation if the tube adjust midrange brightness.

Pro user price of the M-Series absorbers is in the region of $12 per Sabine.

**ACOUSTIC SCIENCES CORPORATION**

**For additional information circle #229**

### DOLBY UNVEILS XP SERIES OF COST-EFFECTIVE MULTITRACK NR UNITS

The XP Series contains up to 24 channels of Dolby A-type noise reduction, and is identical in performance to the SP Series. Improved engineering and manufacturing efficiency permit the 24-track XP Series to be offered at $14,950 — a price 30% less than its predecessor.

Among the XP's cost-effective engineering changes are a new power supply (Model PS3) and dedicated NR circuit design rather than carrier card/plug-in assemblies. The XP also features detented calibration trim controls, discrete FET noise reduction control switching, and individual channel hard-wire bypass.

Dolby will continue to produce SP units in limited quantities at a price of $22,500 for 24 channels.

**DOLBY LABORATORIES, INC.**

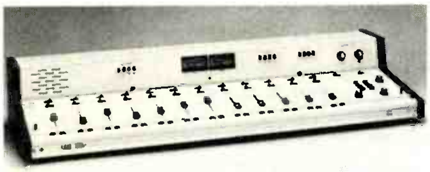
**For additional information circle #230**

### NEW 12-CHANNEL UREI 1690 BROADCAST CONSOLE

The new 1690 Series includes three different models that offer a choice of mixer controls. The Model 1691 is available with rotary conductive-plastic attenuators; the 1692 with Shallco precision-stepped rotary attenuators; and the Model 1693 features Penny & Giles straight-line attenuators.

The signal-to-noise ratio of the microphone channel, from input to console output, is quoted to be better than 74 dB with -50 dBm input and +4 dBm output, or more tham 90 dB referenced to maximum output. At full output level of +24 dBm into 600 ohms, the THD of both Program and Audition channels is quoted to be less than 0.25% over the range of 30 Hz to 15 Hz.

Each mixer position has two inputs selectable with a rocker switch. In addition, three banks of four pushbutton switches may be connected to any mixer input for use with additional sources such as remote or network feeds. An overload indicator LED is located between the VU meters and its threshold can be internally adjusted to alert the operator that a downstream device, such as an STL, may be clipping.



All channel on/off switching is performed by FET switches activated by illuminated pushbuttons; extra switch contacts may be used for activating cart machines, turntables or similar equipment.

The 1690 Series Consoles are supplied with one mono, transformer-isolated mike input pre-amp, and 11 stereo-line input pre-amps.

A four-position pushbutton selector connects Program, Audition, Air or an external input to an internal 8-watt stereo power amp. Another four-position selector sends Program, Audition, Air or Cue to an internal one-watt amp and two stereo phone jacks. A cue loudspeaker with its own amp is built in, and is automatically muted whenever Mute Buss #1 is activated.

**JBL PROFESSIONAL**

**For additional information circle #231**

### AURATONE NOW AVAILABLE WITH FABRIC GRILLE AND SHIELDED MAGNET OPTION

The 5C Super-Sound-Cube close-field mix-down monitors can now be supplied with snap-on black stretch fabric grilles in place of the former foam grilles. In addition, for those who need to position them close to video monitors, the 5C is also available with a shielded magnet structure that is said to minimize deflection of the picture tube image.

Frequency response of both standard and shielded drivers is quoted to be nearly identical, with audible response covering the range from 50 Hz to 17 kHz; on-axis response is ±3.5 dB from 200 Hz to 14 kHz, and power rating 30 watts RMS (60 watts peak at 150 Hz) into 8 ohms.

The standard version lists at $99 per pair, and the shielded-magnet model for $114 per pair. Full-range shielded magnet drivers are also available separately for $31.95, each for users that wish to retrofit older, standard-driver cubes.

**AURATONE CORPORATION**

**For additional information circle #232**

### APHEX INTRODUCES AURAL EXCITER CHIP FOR OEM LICENSING

The basic Aural Exciter circuitry is now available on a monolithic chip dubbed the MAX (Monolithic Aural Exciter) circuit, and can be supplied on a licensing basis for use in a variety of products.

The first licensees of the new chip are: **AKG**, Munich, Germany; **Gentner Engineering**, Midvale, UT; and **Numark Electronics Corps.**, Edison, NJ. Each company is using the circuit for uniquely different products; the AKG Aphex Magic is an automotive add-on for stereo systems to restore the highs lost in noise-reduction systems, and to make the highs more audible over driving noise. Numark is using the MAX in its Proline mixers, while Gentner uses it in the EFT-1000 Extended Frequency Transceiver to restore high frequencies lost on telephone lines used in remote broadcasts.

Compared to discreete Aural Exciter circuits, the MAX chip is said to offer improved performance, including a greatly improved drive window for input level tolerance. The process is claimed to restore natural brightness and presence, improve intelligibility, increase perceived loudness without changing actual gain or EQ.

**APHEX SYSTEMS, LTD.**

**For additional information circle #233**

### PRISTINE ANNOUNCES AUTOMATED RECORDING STUDIO MANAGEMENT SYSTEM

Described as the first microcomputer business system for the small- to medium-size recording studio, The Recording Studio Management System includes studio bookings, work orders, inventory control, billing, accounts receivable and payable, tape library management, general ledger and financial reporting.

The software package is completely integrated and compatible with the most popular microcomputers, including MS.DOS sys-

# New Products

tems and the Apple Macintosh. The system is available in both floppy diskette and hard-disk versions, and custom modifications are available.

**PRISTINE SYSTEMS, INC.**

**For additional information circle #237**

## TRISTECH FX-200 DUAL-CHANNEL NOISE GATE

Each channel has control over both threshold level and release time. Attack time is preset, and is said to be more than fast enough for use with drums and percussion. Release time is variable from 10 milliseconds to two seconds.


A key input and switch is provided for special effects. The unit may be used with any signal level, from -10 to +4 or +8 dB with unbalanced in and out. Gate operation is indicated by a red "signal-on" LED.

The FX-200 is a stand alone unit that may be used individually, or mounted in a 19-inch rackmount. Each self-contained unit comes complete with an external power transformer for low hum and noise specifications. Each unit measures approximately 7 by 2 by 4 inches, and is constructed in a heavy-gauge aluminum case. The FX-200 will carry a suggested retail price of $169.95.

Other units in the FX Series include the FX-1000 level matching box, FX-600 compact stereo power amplifier, and FX-DRM, a dual-rack mounting kit. Future products will include the FX-300 compressor/limiter, FX-800 stereo phono pre-amplifier, FX-400 Feedback Killer, and FX-1000 Stereo Synthesizer.

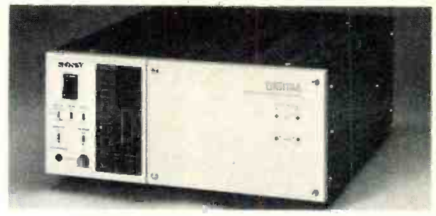
**TRISTECH AUDIO, INC.
AUDIO ENVELOPE DIVISION**

**For additional information circle #238**

## SONY PCM-1630 DIGITAL PROCESSOR AND DMR-4000 U-MATIC

The new PCM-1630 is a refined version of the PCM-1610 first introduced in 1981, and uses newly developed analog and digital filtering techniques, resulting in significant improvement in audio quality, Sony claims. Fully compatible with the -1610 format, the new unit offers the same 16-bit linear quantiization and switchable sampling rate of 44.1 kHz or 44.056 kHz, with a quoted dynamic range of more than 90 dB, harmonic distortion of less than 0.5%, and "unmeasurable" wow and flutter. Direct digital to digital dubbing is also carried over from the PCM-1630's predecessor, as is electronic editing when used in conjunction with the DAE-1100.

Among the unit's new features are peak level meters with peak-hold mode, and an optional Read After Read (RAR) board for improved error correction on playback using the new DMR-4000 recorder. An optional digital I/O board makes teh PCM-1630 compatible with AES/EBU standards. The PCM-1630 weighs 57.5 pounds — about 25 pounds less than its predecessor.



The DMR Digital Master Recorder was developed with the PCM-1630 as a transport dedicated to audio processing. Special features include Read After Write (RAW) and Read After Read (RAR) capabilities, automatic head cleaning and a built-in SMPTE timecode generator-reader. Read After Write (RAW) is now possible because of additional play heads that have been built into the reader's head drum, enabling simultaneous playback upon recording. Read After Read (RAR), allows simultaneous playback of digital data from the DMR-4000's regular record/play and confidence heads. The two digital ouput signals are then compared in an optional board, and dropout errors detected and corrected, resulting in improved accuracy. An eight-pin connector for composite digital and timecode I/O is said to simplify connection to a PCM-1630/DAE-1100 system.

The DMR-4000, available in February 1986, will carry a pro-user price of $14,000, while the PCM-1630 has a pro-user price of $19,000.

**SONY CORPORATION OF AMERICA**

**For additional information circle #239**

## AEG-TELEFUNKEN M-20 AND M-21 SERIES TAPE MACHINES NOW AVAILABLE IN U.S.

The M-20 Series can be supplied in standard two-track or center-track SMPTE timecode configurations. Standard features include complete microprocessor control of all functions; a six-position programmable and stored level, equalization, and bias alignments for four tape speeds (including NAB or proposed AES/CCIR equalization and optimization for three tape formulations at each speed); and external synchronization capability.

The design incorporates a cast-steel deck and modular circuitry, and special ergonomic layout of tape path and controls for ease of operation amd editing.

The M-21 series have been developed for less sophisticated applications requiring only two-speed operation, manual audio alignment, and no synchronization possibilities.

Both Series utilize a unique Amorphous

Metal head design presently available only from AEG. Manufactured by a special process resulting in a previously unattainable hardness, the heads are claimed to provide a lifetime expectancy equal to that of the machine. Their butterfly core construction is said to substantially reduce interchannel crosstalk, and wide profile yields exceptional low-frequency response at higher speeds.

**AEG CORPORATION**

**For additional information circle #240**

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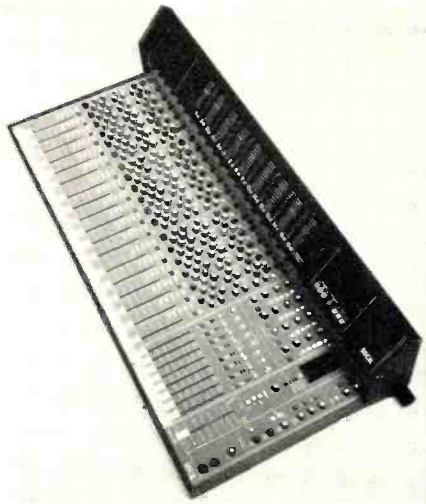
### SECK 1882 PORTABLE MIXER FROM CONNECTRONICS

Latest in the range of SECK portable consoles, the Model 1882 has 18 inputs that can be switched between a mike, line or tape input. All inputs are balanced, and the microphone input has 48-volt phantom power that is switchable at the power supply. Input amps provide extended range from -55 to +10 dBm, with a full 20-dB margin of headroom.

Full in-line monitoring facilities, which provide a stereo monitor mix for multitrack recording and enable the extra facilities to be switched into the signal channel, offer three extra effects busses during mixdown. In addition to the three effects busses available in the monitor section, there are three further effects busses in each input channel, together with a three-band EQ that includes a sweepable midrange section.

To ensure accuracy, a stereo in-place solo facility is provided on all inputs and auxiliary returns; solo is also available on the sub-groups and auxiliary sends.

Eight sub-groups are switchable to 16 track outputs, giving full compatibility with 16-tracks without continuous patching. Insert points are provided on all sub-groups and input channels, allowing dedicated signal processing or feeds at each of these points.



Metering is via LED bargraph meters, and are provided with a peak-hold feature. Monitoring is provided and allows switching between channel monitors, left and right master output, or a two-track during mixdown.

The SECK 1882 has a suggested retail price of $3,995.

**CONNECTRONICS CORPORATION**

**For additional information circle #241**

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### STUDIO TECHNOLOGIES RCU-1 FOR MTS BROADCASTING

The RCU-1 Recognition Control Unit is designed for use in conjunction with MTS television broadcast operations. It determines and displays the mono/stereo status of broadcast audio programming, and automatically switches a stereo simulator into the on-air broadcast chain upon recognition of mono.

By employing VCA-based cross-fading circuitry, the transition from true to simulated stereo is said to be smooth and inconspicuous to the listeners. Extensive circuitry is used to allow ±45 degrees of phase error (at 1 kHz), and channel level difference of 10 dB to still be recognized correctly.

The RCU-1 can also be used as a dedicated mono/stereo recognition device in a television broadcast or production facility. Prior to broadcast, videotapes can be monitored to determine the actual status of the audio channels present.

Designed as a companion device for the AN-2 Stereo Simulator, the RCU-1 is also compatible with stereo simulators built by other manufacturers.

List price for the RCU-1 is $1,200.

**STUDIO TECHNOLOGIES, INC.**

**For additional information circle #242**

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**For additional information circle #243**

# New Products

## FOUR DESIGNS ANNOUNCES NEW CONTINUITY TESTER

Testing 1-2-3 is a pocket-sized unit that checks both balanced and unbalanced cables for continuity, shorts and phase, and instantly displays the results of the test via three LEDs. A complete test takes less than 10 seconds.


The device is housed in a rugged ABS case measuring 3¾ by 2⅝ by 1¾ inches, and weighs 7.8 ounces with standard 9-volt battery. Suggested list price of the Testing 1-2-3 is $49.95

**FOUR DESIGNS COMPANY**

For additional information circle #244

## FOSTEX UNVEILS E-2 AND E22 TWO-TRACKS

The new tape machines feature totally microprocessor controlled transports *and* record/play logic, so they can run computer driven edit decision lists. They also feature center-track for SMPTE timecode as standard, and a synchronizer port.

In addition, the E-2 quarter-inch and E-22 half-inch recorders can be used with Fostex 4050 MIDI Synchronizer/Autolocator to lock up drum machines and synthesizers to SMPTE timecode; automatic programmable punch in/out; selectable pre-rolls to 99 seconds; zone limits; record selects; locate to the measure, bar and beat (as well as time); quantize all MIDI Clocks; and 10-position autolocate.

**FOSTEX CORPORATION OF AMERICA**

For additional information circle #245

## DOD DIGITECH RDS 3600 DIGITAL DELAY

Featuring a full seven-second delay that can be triggered by drum machine, the RDS-3600 also offers footswitch control, plus speed, width and delay functions in flange mode from 1.5 to 14 milliseconds; chorus from 6 to 55 milliseconds; double from 50 to 450 milliseconds; plus echo covering 200 to 1,800, 400 to 3,600 and 800 to 7,200 milliseconds.

LEDs are provided for delay kill, headroom and delay time, while a two-second mute prevents unwanted noise when switching delay time.

Repeat hold can be activated at either the front or the back of the unit, with "feedback" option available, in- or out-of-phase.

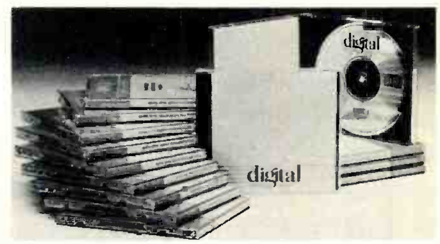
The RDS 3600 retails for $399; the RDS-1900 for $299; and the RDS 900 for $259.

**DOD ELECTRONICS INTERNATIONAL**

For additional information circle #246

## DIGITAL PRODUCTION LIBRARY LAUNCHES NEW MUSIC LIBRARY ON COMPACT DISC

The new library includes only material that has been recorded since late 1984 specifically for this release; all materials have been mixed and mastered specifically for the Compact Disc medium. It is available for a $3,000 licensing fee, including all materials and one-year usage rights, directly from the company. The library is also available to radio and TV broadcasters, but licensing costs for these subscribers vary according to market size.




As a promotion, the company is offering a programmable CD player at no additional charge to the first 500 subscribers to the library. Product demonstration materials and brochures are also available.

Most themes in the library offer long versions of approximatley two minutes, often longer; these are designed for easy edit looping. There are typically six or seven cuts of major themes. The library also includes numerous incidental cuts, ranging from a tympani roll to minute-length selections.

Digital Production Library chose CD as its method of distribution because of the "extraordinary fidelity and immunity to wear of the medium." Programmable CD players permit instant access to cuts, quick scanning of cuts, looping and other production work-savers; the net result is described as dramatically increased productivity. Tape copies of CD masters boast a level of quality that can equal or exceed tape masters; they are also far superior to tape copies of vinyl masters, the company says.

**DIGITAL PRODUCTION LIBRARY**

For additional information circle #247

## APHEX INTRODUCES LOWER-PRICED AURAL EXCITER

According to Aphex president Marvin Caesar, the low-priced Type C unit has better sound characteristics than the Type B, which the new model replaces. A new MAX (Monolithic Aural Exciter) integrated circuit makes the lower price possible, Caesar says.

"The Type C uses the same principle as the more expensive Aural Exciters used on thousands of hit albums, concerts and movies, and on the air on hundreds of radio stations worldwide," Caesar comments. "As with all Aural Exciters, it restores *natural* brightness and presence, makes your sound *real* again."

The Type C has two independent channels that can be used for stereo with matching settings, or separately with different settings. Each channel is line-level in/out, with the nominal operating level -10 dBm. Connections are made via standard phone jacks and/or RCA jacks on the back panel.

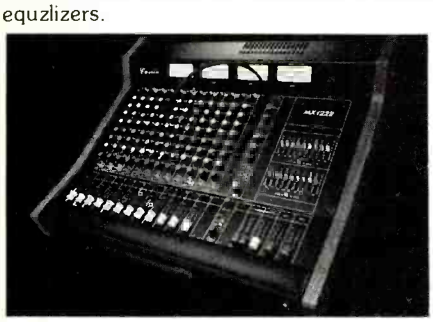
**APHEX SYSTEMS LIMITED**

**For additional information circle #249**

**CARVIN MX-22 SERIES CONSOLE**

Available in 6, 8, 12, 16 and 24-channel formats, the MX-22 Series consoles feature ultra-low noise differential mike pre-amps with mike/line switching (typical output-noise performance is a quoted -90 dBm); 48V phantom power; three-band equalization with sweepable mid-range; two monitor sends per channel; one effects send (with built-in Hammond three-spring reverb system); stereo pan control, and priority solo switching.

The master section features two-track outputs, Mono-master output. BNC Littlelite, full talkback facilities, Monitor #1 and #2 outputs, and a pair of nine-band graphic equzlizers.



The majority of inputs and outputs are +4 dB/-10 dB. Quoted input noise is -128 dBv, crosstalk -65 dB at 1 kHz and -55 dB at 10 kHz, CMR -75 dB at 1 kHz, and THD typically less than 0.02%, 20 Hz to 20 kHz.

Pro-user prices range from: MX-622P, including built-in amp, $799; to MX-2422 (without amp), $1,999.

**CARVIN CORPORATION**

**For additional information circle #250**

**ATR-60 SERIES OF TAPE MACHINES FROM TASCAM**

The new Series is offered in six versions: ATR-60-2T two-track format with center-track timecode on ¼-nch at 15 ips/7.5 ips; ATR-60-2N two-track NAB stereo, on ¼-inch at 15 ips/7.5 ips; ATR-60-2D two-track DIN stereo, on ¼-inch at 15 ips/7.5 ips; ATR-60-2HS two-track stereo on half-inch at 30 ips/15 ips; ATR-60-4HS four-track NAB format on half-inch at 30 ips/15 ips; and ATR-60-8 eight-track IEC format on half-inch at 15 ips/7.5 ips.

Each model is built with the transport in one chassis, and the electronics in another, a

# New Products

design that enables the transport and electronics to be mounted in a standard rack, in a portable rack case, or in a console rack.

The transport is built on an extra heavy duty chassis that is said to assure stable alignment and the utmost accuracy in tape motion. Microprocessor control ensures smooth, fast, and accurate tape motion by commanding a pair of direct-drive reel motors and a Phase-locked Loop servo capstan motor. Even in synchronized lock-up to other audio transports, film chains or video systems, the ATR-60 has sufficient accuracy to keep in step with a busy work schedule, Tascam says.

**TASCAM**

**For additional information circle #255**

### NEW SCOTCHCART II LONG-LIFE BROADCAST CARTRIDGE FROM ITC/3M

The new premium-grade NAB cartridge is said to offer a new, higher output, low noise, lubricated tape for recording at high levels without performance loss, and five times the average life of its nearest competitor.

ScotchCart II cartridge uses no pressure pads, resulting in the elimination of problems with tape steering, wear and excessive audio side-band noise. Utilizing a new temperature stable, non-rotating hub, the cart is said to eliminate the mechanical irregularities of rotating hubs found in conventional cartridge designs. Concave guides position the tape, allowing the cart machine to handle the critical guidance. A patented dynamic-tension control system insures proper tape-to-tape head contact, provides constant tape tension, and controls tape looping inside the cartridge.

Tape continues to exit naturally from the hub center, instead of twisting and curling up and over the pack like conventional cartridge designs; this feature, combined with an adjustable cam to control the tape loop, is said to further help ensure maximum cartridge life.

**INTERNATIONAL TAPETRONICS CORPORATION/3M**

**For additional information circle #256**

### MODEL Q5000 PROGRAM EQUALIZER FROM RLA

The new unit is a five-band, stereo equalizer equipped with special features to facilitate sub-bass control. Particluar attention has been paid to the design of center frequencies and bandwidths to optimize the Q5000's effect on the tonal quality of program material, the company says. Boost/cut ranges are modest at +8 dB, in keeping with the requirement to augment source material, not to destroy it. This also maximizes the sweep range of the controls, allowing precise control of subtle, but important changes.

A special Sub-Bass EQ loop is provided, which automatically disconnects the loop. Anytime the loop is engaged, Band-1 becomes either a 50 Hz sub-bass equalizer or a main line 120 Hz equalizer, to allow the operator to switch back and forth between deep bass control of the sub-woofers and low-end control of the mains, as the program material dictates. The unit is designed to complement the Model X3000A active crossover.

**RLA INTERNATIONAL, LTD**

**For additional information circle #257**

### CARROLLTRONICS MULTI-AMP LINE AMPLIFIER

The new unit can be configured as 12 independent gain stages, or as a distribution amplifier with any combination of outputs driven from common inputs; selection is via internal links, Commonly-used D/A configurations would be two sets of six outputs, three sets of six outputs, three sets of four outs, etc.

Studio applications include raising the –10 dBV level outputs of cassette machines, synthesizers and other semi-pro equipment to +4/+8 dBm studio level. Video facilities can also benefit by leveling all VCR outputs before switching, as well as using the distribution amp configuration.

Each channel of the rack-mount unit is provided with unbalanced ¼-inch phone and RCA input jacks (in parallel on the front panel), screwdriver-adjust gain control, and active transformerless-balanced output on the rear panel. Outputs are provided on three-pin XLR-type connectors.

**CARROLLTRONICS**

**For additional information circle #258**

### NEW SBX-10 SYNC BOX/CONVERTER FROM ROLAND

The SBX-10 reads click, MIDI, DIN-sync (24 or 48 pulses per quarter note) or Time-Base (24, 48, 96 or 120 pulses per quarter note) signals, and then simultaneously synchronizes all devices connected to its two MIDI-Out connectors, two DIN-sync jacks and two Time-Base Out jacks. Up to four different Time Bases can be synchronized simultaneously. External devices can be started at any point in a song (even between beats), and they will always lock in precisely to the beat, Roland claims.

The unit features a memory capacity of up to 1,024 quarter notes. Programming can be done either by tapping the Tap button or by feeding audio click signals to the Click-In jack. The connected drum machines, sequencers, and other devices can then be completely synchronized with expressive, subtle changes in tempo.

Suggested retail price of the SBX-10 is $325.

**ROLAND CORP US**

**For additional information circle #259**

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1.(a) Title of publication. **RECORDING Engineer/Producer**
(b) Publication No. **34163**
2. Date of filing: **October 1985**
3. Frequency of issue: **Bi-monthly**
(a) No. of issues published annually: **Six**
(b) Annual subscription price: **$24.00**
4. Address of known office of publication:
**1850 No. Whitley Ave., Hollywood, CA 90028**
5. Address of the headquarters of general business offices of the publisher (not printer).
**1850 No. Whitley Ave., Hollywood, CA 90028.**
6. Names and addresses of of Publisher and Editor: **Publisher, Martin Gallay, 1850 No. Whitley Ave., Hollywood, CA 90028; Editor, Melvin Lambert, 1850 No. Whitley Ave., Hollywood, CA 90028.**
7. Owner: **Gallay Communications, Inc., Martin Gallay**
8. Known bondholders, mortgagees, and other security holders owning or holding 1 percent or more of total amount of bonds mortgages or other securities: **None.**
9. For completion by nonprofit organizations authorized to mail at special rates (Section 423.12 DDM only): **Not applicable.**
10. Extent and nature of circulation:

| | Average number copies each issue during preceding 12 months | Actual number copies of single issue nearest to filing date |
|---|---|---|
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| B. Paid and/or Requested Circulation | | |
| 1. Sales through dealers & carriers, street vendors and counter sales | 2.500 | 2.500 |
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— continued from page 28 . . .

BBC Television Centre, London, will each receive two 5000 consoles: a main console with 36 channels and eight VCA Control Groups; and a "Gram Ops" desk with 10 stereo channels for premixing stereo sources from disks, tapes and CDs.

• The first U.S. delivery of the new 5000 series will be to NBC Radio News, which will be receiving a 40-channel version with a mixture of mono and stereo facilities plus eight Independent Main Outputs (IMOs). The console will be placed into a new studio now under construction in Burbank, CA, and which will gather feeds from many different sources, process them for live distribution, and then re-transmit them via satellite to the NBC Radio News Network master control room in New York.

• Another SL-5000M Series has been ordered by Tokyo Genzosho, the film and video production facility of the Eastman Kodak Far East Company in Tokyo, Japan. The console will be fitted with 16 mono channels and eight stereo subgroups, and is intended for use in video pre-production for VCR and LaserDisc mastering.

All six of the initial SL-5000M Series consoles will be equipped with SSL's Instant Reset Computer, which provides instantaneous switching between any of 48 "stores," each of which records the status of all switches throughout the system.

• The company also reports receiving its 300th order for an E-Series console. "Number 300 will be an SL-4000E Series with 40 channels and SSL Studio Computer," according to Andy Wild, marketing VP of American operations. "It has been purchased by Geordie Hormel's Village Recorders for use in their Studio A in West L.A." The console will also represent the twentieth SSL system to be purchased by a Los Angeles facility.

### SMPTE TO STUDY STANDARDIZATION OF 30-FRAME FILM FOR DISTRIBUTION OF THEATRICAL AND TELEVISION PRINTS

The Society of Motion Picture and Television Engineers is organizing a study group to consider the feasibility of changing the frame rate of films produced for theatrical and television distribution. The current 24 fps rate was established by the Society in the late Twenties with the advent of sound recording, and remains today the world standard for theatrical projection. The group will examine a 30-frame rate that offers improvements for both theatrical and TV film applications. The higher rate would be compatible with the existing 30-frame television standards, and the proposed 60 Hz HDTV world standard, supported by the U.S. and currently under consideration by the CCIR. Its adoption would also offer advantages for theatrical projection, including improved sound, reduced flicker and improved picture quality.

The 30-frame concept is not new, however; many commercials are photographed at this rate, and a number of special systems have taken advantage of the improvements offered by an increased frame rate.

### COMPUSONICS AND AT&T ANNOUNCE JOINT MARKETING AGREEMENT FOR DIGITAL AUDIO TRANSMISSION TECHNIQUES

The one-year agreement calls for the two companies to jointly promote CompuSonic's patented Telerecording equipment and AT&T's Switched 56 Service, for use together in sending digital audio signals over long distances.

The Telerecording technique permits music to be digitally transmitted and received between remote data bases, using special telephone lines as the transmitting channel. The technology is an optional feature of CompuSonic's patented hard-disk-based digital audio recording and playback system, which is expected to be in service commercially later this year.

According to CompuSonics president David Schwartz, "Consumers will literally be able to phone in their music pur-



# News

chases to the record dealer, pay for the album with a credit card, turn their CompuSonics consumer recording system to the 'telerecord' mode, and receive the album over the telephone line on to a specially formatted 5¼-inch floppy disk."

## SSL CONFIRMS DIGITAL RESEARCH; ANNOUNCES RELOCATION OF INTERNATIONAL HEADQUARTERS

Colin Sanders, founder and Managing Director of Solid State Logic, recently announced plans for a major expansion of the company's international headquarters at Oxford, England, and confirmed rumors that SSL is engaged in an extensive digital research and development program. The company plans to move its administrative, research and training centres from Stonesfield, where the company has been based for the last 15 years, to a new site in the nearby town of Woodstock. When completed, SSL's new international headquarters will comprise over 62,000 square feet, and house more that 300 staff.

Turning to the future, Sanders confirmed that SSL has been "quietly engaged in digital research for the last several years," and that this research will accelerate as the company moves to its new site.

"The goal of this research is an entirely digital SSL Studio System. I hasten to add that this is *not* a product announcement; we are not yet taking orders for the system, nor can I confirm exactly when it will be available or what it will cost. We are working not to a specific date or price — we are working for a performance specification that will satisfy the most critical listener."

However, Sanders has revealed that the SSL Digital Studio System will be based on a proprietary 24-bit digital audio processor; that it will incorporate integral audio storage and editing capabilities; and that it "will be one of the most powerful computers ever built."

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— News Notes —

• **Ruby (Moore) Hunsberger,** co-founder and Chairman of the Board for **Crown International** since 1979, has retired from the daily decision making of the company. She co-founded Crown 35 years ago with her former husband, the late Clarence Moore; when Clarence died in 1979, she assumed Chairmanship of the Board of Directors. Her son, **Clyde Moore** has been appointed to take her place as Chairman of the Board's Executive Committee.

• **Solid State Logic** has appointed **Audio Intervisual Design,** of Los Angeles, as its Special Consultant for large-scale systems applications. According to Andy Wild, who heads SSL's West Coast operations, "Audio Intervisual's expertise in studio design and equipment interface neatly complements Solid State Logic's total systems approach to console and studio computer design and applications. We are looking forward to working with Chris Stone and his staff on a variety of major film and video projects."

• **Mel Foster Technical Sales,** Edina, MN. has been appointed the new sales representatives for **Community Light and Sound** in the North Central region The company specializes in audio and sound reinforcement product sales, and will handle the entire Community product line in the states of Minnesota, North and South Dakota, and western Wisconsin.

• **Valley People** has appointed the following representatives for the company's product line: **Darmstedter Associates, Electro-Acoustic Marketing, Wilson Audio Sales, Bencsik Associates, Dobbs Stanford Corporation, YoreCo, RM Associates,** and **Radon and Associates.**

• **Adams-Smith** has set up a U.K. subsidiary to provide end-user and distributor support for Europe. The new firm will be managed by **David Godsmark,** formerly of Pye TVT, Ltd. "As the number of System 2600 installations in Europe increases," says Andrew Simon, company president, "we want to make sure that our customers have access to European-based sales and service. Also, with the addition to System 2600 of the 2600 A/V Double-System editor, it is important to us to have highly competent technical support available to our distributors throughout Europe."

• **FM Acoustics USA** have moved to the following address: P.O. Box 854, Benicia, CA 94510. (707) 745-4444.

• **Trident Audio Developments, Ltd.** has moved to new and larger premises. The company's new address is: Trident House, Rodd Industrial Estate, Govett Avenue, Shepperton, Middlesex TW17 8AQ, England. Telephone: Walton-on-Thames 224665. Telex: 881392 TRIMIX G.

• **Audio Kinetics** has released details of several more tape-machine interfaces for the Q.LOCK synchronizer, including the Studer A820, Tascam MS-16 16-track and Otari MX-70 16-track; further plans include interfaces for the Nagra T-Audio and Otari MTR-20, amongst others.

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## Swiss Audio: Technical Evolution


# On adding time-saving production features to a proven audio recorder design.

The updated PR99 MKII, now offering a microprocessor controlled real time counter, address locate, zero locate, auto repeat, and variable speed control, can improve your audio production efficiency. And, as before, it's built to meet strict Studer standards for long-term reliability.

Welcome to real time. The PR99 MKII's real time counter gives a plus or minus readout in hours, minutes and seconds from −9.59.59 to +29.59.59. Counter error is less than 0.5%, and the microprocessor automatically recomputes the time displayed on the LED counter when you change tape speeds.

Fast find modes. Press the address locate button and the PR99 MKII fast winds to your pre-selected address, which may be entered from the keyboard or transferred from the counter reading. Press zero locate and it fast winds to the zero counter reading. In the repeat mode, the PR99 plays from the lower memory point (zero or negative address) to the higher point, rewinds to lower point, and re-activates play mode for a continuously repeating cycle.

Pick up the tempo? When activated by a latching pushbutton, the front-panel vari-speed control adjusts the nominal tape speed across a −33% to +50% range. The adjustment potentiometer is spread in the center range for fine tuning of pitch.

Future perfect. The PR99 MKII also offers a serial data port for direct access to all microprocessor controlled functions.

Much gained, nothing lost. The new MKII version retains all features of its highly regarded predecessor, including a die-cast aluminum chassis and headblock, balanced and floating "+4" inputs and outputs, self-sync, input mode switching, and front panel microphone inputs.

European endurance. Designed and built in Switzerland and West Germany, the PR99 MKII is a product of precision manufacturing and meticulous assembly. Every part inside is made to last.

To discover more about the world's most versatile and dependable budget-priced recorder, please contact: Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

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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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If you're going omni. Our new SM90 is identical in appearance to the SM91 and just as rugged.

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