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AUDIO PRODUCTION FOR BROADCAST
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Volume 14 — Number 6

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— The Cover —
Control room interior at **Rocshire Studios**, Anaheim, California. A feature article detailing the intricate reconstruction process to improve sound isolation begins on page 96.
Photography by **Barry Levine**.

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DIRECT-TO-DIGITAL

from: **Bob Burr, president**
QL Mobile Recording
Coral Gables, Florida

I was pleased to read in October's "News, Letters, and Views" column a letter by Jeffery Weber concerning recording direct to two-track.

We all know that real men don't eat quiche. I submit that the recording equivalent of quiche is the ever popular multitrack overdub approach to making music.

It didn't happen overnight. We are all guilty of contributing to the collective evolution of what we now call "recording." Have we not lost sight of reality in music? When did the recording business get boring? When we started overdubbing *everything*.

I can't expect you to agree with me if you own, or make your living in, a modern multitrack facility. Your job is to give your clients what they want. Play it safe. Overdub 'til the cows come home, usually one track at a time. You *will* get it right eventually, and at $100+ per hour, who's complaining? Not the studio owner. Not the union musicians.

My feeling is that at least 25% of popular recording should be done live in the studio. Obviously, classical music is rarely overdubbed. Most good jazz happens in real time. Ditto for reggae, folk and new wave. Why not record it in real time? All it takes is three things ... talent, talent and talent.

Perhaps you've read books about the early days of recording, or talked to people who were there. If not, I'll tell you a little story. You see, in the good old days, the musicians actually showed up at the same time, in the same place, and played together. I know it sounds funny, but sometimes something magic would happen as the guys got to playing. On a good day they would really get "into a groove." The signal went directly to a mono or stereo tape machine or, better yet, direct to the disk cutting head. Bingo! A hit record. Press it up.

In these modern times we can, at best, create the illusion that the musicians played together by donning headphones, closing our eyes, and pretending to be there together.

Since we live in the future now, let's take this one step further. What if we record direct to digital (or if you're still scared of digital, half-inch analog). Book time in your favorite world class studio for three days instead of three months. A half dozen or so takes of each tune can be electronically edited into a "perfect take." Disk mastering may be performed repeatedly to achieve dramatic results. Compact Discs, analog and digital audio cassettes are ready to be mastered from a purely digital source. It doesn't get any cleaner than this folks.

Please be careful. Once you've tried it you may find, as I did, that multitracking all night long will put you to sleep. So try recording direct-to-digital. Like I said before, all it takes is three things ...

---

dBv or dBV?

from: **Steven Graham**
The University of Michigan
Public Radio Stations
Ann Arbor, Michigan

This is an addendum to the discussion of dBv versus dBm found on pages 124 and 125 of the October, 1983 issue ["Performing Meaningful Noise Measuerments in Theory and Practice," by Paul C. Buff].

To more clearly distinguish between dBv (0 dB = 0.775 VRMS) and dBV (0 dB = 1.0 VRMS) it seems that dBu and/or dBU are being used quite often instead of dBv. (It is often difficult to distinguish dBv from dBV if it is handwitten, and "dB-small-v" is a mouthful.)

Editor's Note: John Roberts' column "Exposing Audio Mythology," to be found elsewhere in this issue, should further clarify the differences between these often confusing reference levels. *R-e/p* would be interested to hear from other readers regarding the standardization of dBu/dBU to designate levels referenced to 0.775 VRMS, to avoid confusion with dBV (1V).

PRESSURE ZONE MICROPHONES® — ANOTHER VIEW

from: **Stephen F. Temmer**
Gotham Audio Corporation
New York, New York

With reference to the increasing use of Pressure Zone Microphones in recording and production studios, readers of *Recording Engineer/Producer* may be interested in the following translation made by this writer of an article by Eberhard Sengpiel, which first appeared in the May/June 1983 issue of the *Newsletter of the Verband Deutscher Tonmeister e.V.* (Association of German Studio Engineers), and titled "Boundary Surface Microphones; A Citicism of the PZM":

"In recent times, more and more microphones have appeared on the marketplace which one can categorize as boundary surface microphones. Different designations such as boundary-barrier or pressure-zone have been used for them. The first of this series of microphones had the trademark PZM® (Pressure Zone Microphone). The deficiencies of this microphone type were discussed in the AES preprint 1796 (F-5) of May 1981: "The Acoustic Behavior of

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PZM® and Pressure Zone Microphone® are registered trademarks of Crown International, Inc.

Pressure-Responding Microphones Positioned on Rigid Boundaries — a Review and Critique," by Lipshitz and Vanderkooy, University of Waterloo/Ontario (Canada). Here is a short excerpt of the most important points made in this article:

The PZM microphone is a new type of microphone. This kind of microphone is exemplified by the fact that the sound does not reach the membrane directly since the membrane is oriented toward a hard boundary surface. The microphone, so to say, gets in its own way. As a justification for this, we find the false assumption by its developer, that in a direct sound field a front oriented membrane at the front of the microphone's main axis produces a rise of high frequencies. The authors prove that this is not true, and that the frequency response and the polar diagram are very adversely influenced by covering the membrane.

For large and sound impervious boundary surfaces one gets a pressure doubling which results from the coherent addition of the reflected signals combined with the direct signal at the boundary surface. As a result one has a level gain of 6 dB. This, however, is only then true when the surface against which the microphone is mounted is large by comparison to the sound's wavelength. For low frequencies the half-field directional characteristic transitions into an omnidirectional characteristic, which leads to a loss of the pressure build-up, and manifests itself as a 6 dB rolloff below the transition frequency. On top of that, the insufficiently sound impervious hardness of the boundary surface and its finite size have effects which cannot be neglected.

The idea of mounting a microphone to a boundary surface was pursued because comb-filter type interference patterns which appear between the direct and reflected sound wave from a surface might be avoided. The most obvious solution is to place the pressure transducer in the plane of the effective boundary surface or to attach it in such a way, that the distance to the boundary surface is small when compared to the sound's wavelength. For stereo recordings such boundary surface microphones can only be used in AB (widely spaced) stereo. This leads to the well-known adverse affect on the localizability, the phase relationship, and the incompatibility with mono.

The name PZM was derived from the pressure zone which exists only in the most immediate neighborhood of a sound impervious boundary surface. That is an area in which the sound velocity at right angles to the boundary surface tends toward zero. If the surface is small, or the wavefront does not impinge on it at right angles, one gets a progressive acoustic wave with a velocity vector which is tangential to the boundary surface.

Figure 1A shows how in this version of a PZM a pressure transducer membrane is

...xed in such a way that it is opposite the boundary surface, instead of being being in its plane and pointing frontward. From Figure 1B one can see clearly that this placement is acoustically equivalent to a membrane in the plane of the surface, in front of which an obstacle has been placed. The sound cannot reach the membrane except through a circular gap around the edge of this obstacle.

A boundary surface microphone also produces an effect on the relationship between direct and diffuse sound (reverberation) and, as a matter of fact, in favor of the direct sound. As an explanation we are given the following: One imagines a reverberation chamber as shown in Figure 2 which produces a completely diffuse, reverberated sound field. In Figure 2A we find the microphone in this reverberant field as well as in the path of a direct sound field of a sound source with distance d. Now the distance d is kept constant and the microphone is moved until it is in the plane of a boundary surface, as shown in Figure 2B. Because of the coherent addition, the direct sound as well as the diffuse sound pressure are raised by 6 dB at this boundary surface. The diffuse sound (reverberation) will, however, only be received within a solid angle of two steradians (hemisphere), by contrast to the initial four steradians (sphere). For pure diffuseness one obtains a 3 dB loss in diffuse reverberation intensity. As a result, the relationship of direct to indirect sound
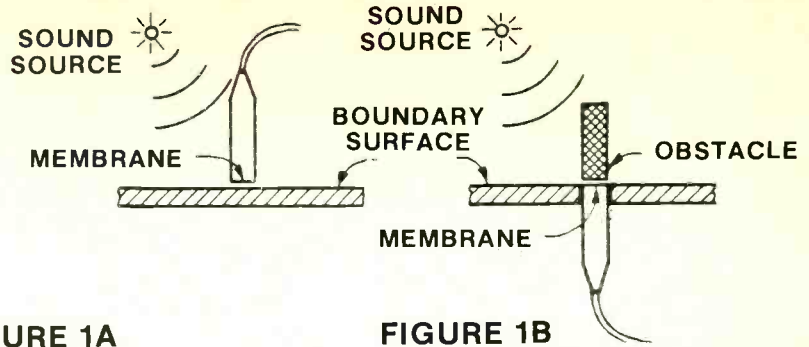
FIGURE 1A　　　　**FIGURE 1B**
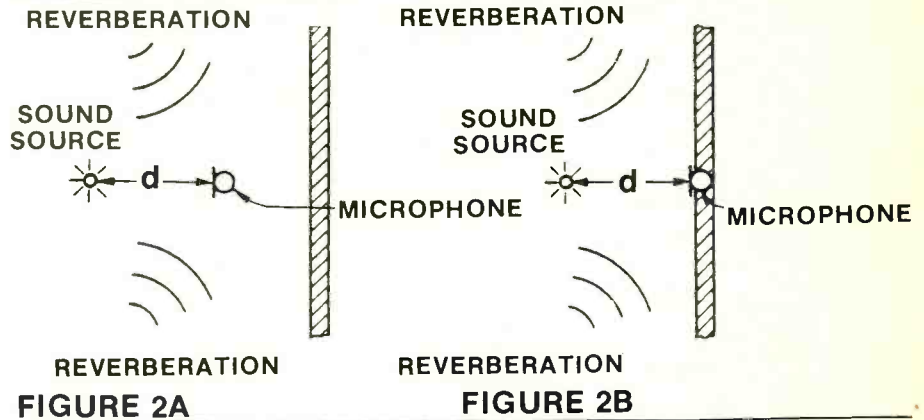
FIGURE 2A　　　　**FIGURE 2B**

will therefore be raised by 3 dB.

By means of a theoretical model and practically using a B&K calibrating microphone 4134 with an effective membrane diameter of 7 mm [0.27 inches], it can be proven and even calculated that the reverse application of the PZM microphone as shown in Figure 1A, is based on false assumptions. Using further diagrams it is easy to see that an obstacle in front of the membrane will lead to standing waves which effect frequency response and polar characteristics in very unequal fashion. Only a small pressure transducer with a membrane diameter of less than 12 mm [0.47 inches] which is in the plane of a sound impervious large boundary surface, will deliver an ideal frequency response and a polar diagram with great integrity over the entire audible range.

These investigations have led to the development of new boundary surface microphones which do not suffer from these errors. The future will show what the applications for this new microphone type are going to be, and whether they will be successful. Comparative investigations about boundary surface microphones made by different manufacturers are not yet available."

Reply from: Bruce Bartlett Microphone Project Engineer, Crown International, Inc.

We agree that flush-mounting a pressure-calibrated microphone in an infinite plane provides a flat frequency response. In fact, at Crown we use a ¼-inch flush-mounted B&K pressure-calibrated microphone as a flat-response reference.

As you mentioned, Pressure Zone Microphones gain 6 dB in sensitivity when mounted on a large surface. The ratio of direct sound to reverberant sound is increased 3 dB, which helps aid clarity by reducing the recorded room ambience.

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LETTERS

Most of the assertions of Lipshitz and Vanderkooy are true as far as they go, but may be misleading. Let's explain:

1. The experiments they did with obstructing a flush-mounted pressure microphone with a baffle are academically interesting, but unrealistic. They made measurements with circular baffles much larger than the size of PZM elements; the smaller the baffle, the higher in frequency the effects occur. Also, rectangular baffles (which PZMs use) have less effect than circular baffles. (Crown also advises against circular baffles for PZM mounting, due to their rougher frequency response.)

When a microphone has an obstruction the size of a PZM cantilever (0.315 by 0.370 inches), the resonant boost occurs above the audible range. Consequently, the frequency response and polar response are not severely degraded by PZM-type mounting. Even if a high-frequency rise does occur, it can be controlled with careful acoustic design.

Interestingly, the diaphragms of many conventional microphones are obstructed intentionally by a Helmholz resonator to boost the high-frequency response.

2. After thousands of hours of usage, the PZM has yet to be demonstrated to have off-axis coloration. It is optimized for uniform polar response below 10 kHz — the most important range for stereo imaging.

3. When a microphone is used in or near a finite baffle, the loss of pressure buildup at low frequencies results in a 6-dB shelving, not a rolloff. This is quite a different effect. For example, the response of a PZM on a two-foot square boundary is down 6 dB at 94 Hz and below. This can be a desirable effect to reduce room runable.

4. It's true that spaced-pair stereo miking provides less-sharp imaging and poorer mono compatibility than near-coincident or coincident miking. However, PZMs can be mounted at the junction of vertical boundaries and the floor. This allows near-coincident stereo miking. PZMs also can be mounted back-to-back on a suspended panel, or on boundary assemblies, for mono-compatible recordings (see Mike Lamm's article, "Realistic Stereo Miking for Classical Recording," in the August issue of *R-e/p*).

5. PZMs do not require a microphone capsule with a pressure-calibrated frequency response. We design PZMs to compensate for any effects of mounting the capsule face-down. It's possible to achieve almost any desired response (including flat) with a PZM by careful acoustic/mechanical design.

We find that mounting the capsule face-down in a cantilever holder offers several benefits:

1. The capsule/cantilever can be removed from the plate. This allows the user to mount the capsule at the junction of multiple boundaries (say, in corners) for greatly increased sensitivity, signal-to-noise ratio, and directivity. Permanent flush-mounting doesn't offer this flexibility.

2. The capsule is better protected from dust and spills.

3. There's an airspace or gap between the capsule and the sound-reflecting plate it faces. By varying the gap parameters, we have an extra measure of control in tailoring the frequency response.

New information about PZM acoustics is gradually becoming available to the public in revised data sheets (dated May '83 or later). Crown also has a new "PZM Theory and Application Guide" which is available free for the asking, and is included with each new PZM.

We highly value subjective user comments since they tell us what parameters are audibly important. Measurements and listening tests don't always correlate. For example, many much-loved studio microphones have off-axis coloration and erratic phase response. PZM measurements do indicate a wide-range, smooth response and very low self-noise; but it's the *sound* of the microphones — and their enthusiastic acceptance — that really counts.

I've used high-quality condenser microphones in situations where they've worked better than PZMs, and was impressed with their beautiful sound. But I've also used PZMs in situations where they sounded better than condensers! It all depends on the application and the taste of the listener. That's why there's a need for many types of microphones. The PZM is not the best mike for every application — no microphone is. But it is a legitimate new tool for the professional recording engineer.

John Roberts' column, "Exposing Audio Mythology," begins on page 19

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For additional information circle #12

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EXPOSING AUDIO MYTHOLOGY

Laying to Rest . . . or at least exposing the false premises upon which they are based . . . some of the Pro-Audio Industry's more obvious "Old Wives Tales"

by John Roberts

In this month's column I would like to shed a little light on the proper usage of Volume Units and Decibels. In the second part we'll take a look at some of the theoretical and practical differences between +4 dB and -10 dB level systems.

On VU and dB and dBm and dBV and . . .

Some of the better advice I received in the course of researching this month's column was that I ought to find something else to wite about; I agree that the subject of reference levels has been the source of much confusion. But please bear with me, because I think this is worth getting straight.

The beauty of the decibel is that we can and do use it routinely for tasks that range from setting equalizers to measuring frequency response, *without* having to know that a decibel is defined as being equal to 10 times the log of the ratio between two given power levels.

$$n = 10 \log_{10} (P1/P2) \ldots \#1$$

By virtue of the ear's logarithmic response to sound levels, the use of decibels allows us to draw useful conclusions about the relative loudness of signals. The operative word here is "relative"; P1/P2 in Equation #1 is simple the *ratio* of the two powers being compared. If you have a scientific calculator handy (and which has log-base 10 — not ln or natural log) you may wish to follow along.

Have you ever wondered why frequency response and crossover filters are often specified by their -3 dB points? Well, it is no coincidence that -3 dB is exactly ½ the power of 0 dB:

$$n = 10 \log_{10} (0.5)$$
$$n = 10 (-0.301)$$
$$n = -3.01 \text{ dB (close enough)}$$

Although decibels are technically limited to describing power ratios, allowances have been made to accommodate available measuring devices. To actually measure power we would have to simultaneously measure the voltage as well as the current within a circuit. Then multiply the two ($P = E \times I$). Much easier said than done.

Using Ohms law ($I = E/R$), we can substitute E/R for I in the power equation, resulting in $P1 = E1 \times E1/R1$, or $P1 = E1^2/R1$ and $P2 = E2^2/R2$. Inserting these new relations for power into Equation 1 we get:

$$n = 10 \log [(E1^2/R1)/(E2^2/R2)] \ldots \#2$$

Now is we assume R1 = R2, Equation 2 further reduces to:

$$n = 10 \log (E1^2/E2^2) \ldots \#2A$$

We can further simplify equation 2A if we take advantage of the relationship, log (x^2) = 2 log (x). Equation 2A now reduces to

the popular form:

$$n = 20 \log (E1/E2) \ldots \#3$$

So far we have discussed the decibel's ability to describe relative levels; however, it is often desirable to define a level absolutely. To do this we may describe a level as being ±n dB from some predefined reference. In audio, the most popular and probably most misused reference is 0 dBm. Quite simply 0 dBm = 0.001 watts. You will notice that dBm is strictly a *power* reference. The most common mistake is to use dBm as a 0.775-volt voltage reference. Specifications thus made will *only* be true for circuits terminated by 600 ohms (0.001 W = 0.775²/600). Perhaps in the good old days when everything was 600-ohms in and 600-ohms out, such casual specification could be tolerated. Much of today's equipment, however, is designed to be used with a wide range of terminating impedances.

Let's take the example of a black box specified for a nominal operating level of 0 dBm, with a maximum input/output level of +20 dBm. For input and output terminations of 600 ohms, we obtain a perfectly reasonable nominal voltage of 0.775 volts and a maximum of 7.75 volts. But should we terminate the output with 20 kohms, that +20 dBm output would have to deliver 44.7 volts to realize the same power as 7.75 volts into 600 ohms!

It is acceptable and probably still useful in transformer-coupled interfacing to spec an output level in dBm, followed by a qualifying impedance in parenthesis, such as +20 dBm (600). However, most contemporary equipment is voltage not power limited, and will put out roughly the same voltage into a wide range of impedances. Specifying the output relative to a voltage reference will provide more meaningful information.

The term "dBV" is used when referencing voltage ratios independent of impedances. Proper usage of dBV requires stating an *actual* reference voltage. For example, the preceding example could have been stated as +20 dBV (0.775V). Popular usage has linked 0 dBV to 1 volt, and dBv, dBu, and dB.7V to 0.775-volt references but, to my knowledge, these popular forms are not official. *When in doubt spell it out* — it is always better to have more information than you need to evaluate a specification, than not enough.

The assumption that "R1 = R2" from between Equations 2 and 2A is fundamental to the use of dB for describing voltage ratios. If R1 doesn't equal R2, the power ratio will *not* be a simple function of the voltage ratio, and only Equation 1 should be used. To see why, take the example of a microphone step-up transformer. While

the secondary voltage is increased, the output current decreases a like amount for a roughly equal power at both the input and output. Thus the gain of any transformer, no matter what the turns ratio, is 0 dB. (Note: there will actually be slight losses for a small -dB gain).

The VU

Now that I've beaten the poor decibel to death, lets take a look at the "Volume Unit." The VU Meter, or more properly "Standard Volume Indicator," is a very tightly defined (USA C-16.5 1961) moving-coil voltmeter. While the actual specification is quite long and detailed. the more critical definitions are: (1) an in-use sensitivity of 0 VU = 1.228 volts; (2) the VU or unit of measure is equal to 1 dB; and (3) a very specific overshoot and signal integration ballistic.

Uniform sensitivity between VU meters facilitates the interfacing of various equipment by providing standard operating levels to design around. The uniform ballistic allows users to reliably interpret what is essentially an average-responding meter when driven by complex (speech and music) waveforms, even when dealing with equipment from different manufacturers.

However, *true* VU meters are relatively expensive (about $50), and not commonly found on lower-cost equipment. The majority of meters used to monitor audio levels today don't even try to meet the 20-year-old standard — thus the high price for the few that do try.

This inconsistency between "true" VU meter ballistics and their lower-cost brethreren is made somewhat less problematic by the proliferation of peak indicators into even budget-priced equipment. With peak information readily available, the user is less likely to grossly misjudge a signal's dynamics. In fact, widespread use of digital electronics will make the display of peak information somewhat more important than average levels for all except loudness balancing, since digital overload is analogeous to electrical clipping, and rather abrupt.

Today, with the cost of electronics dropping faster than their mechanical counterparts, it might be worth considering building RMS computing into dedicated loudness or volume meters, and rely upon peak indicators for headroom information.

+4 versus -10 dB Systems

As anyone who has ever tried to hook up a piece of hifi gear into a studio chain has quickly learned, all 0 VU levels are *not* created equal. It's safe to assume that a $99 cassette deck will not feature true VU meters. In all fairness to USA C16.5 1961, there is only *one* 0 VU, so let's call nominal system operating levels their "0 dB point."

The more expensive consoles designed for +4 dB output levels usually offer those real VU meters, and thus have a nominal operating point of 0 dB = 0 VU = 1.228 volts. The identification as "+4 equipment" is a carry over from the implied 600-ohm terminations, with 1.228 volts into 600 ohms delivering a power level of +4 dBm.

Equipment referred to as "-10 dB gear" is not usually designed to drive 600 ohm levels, and has a nominal system operating level of 0 dB = -10 dBV (1V) = 0.316
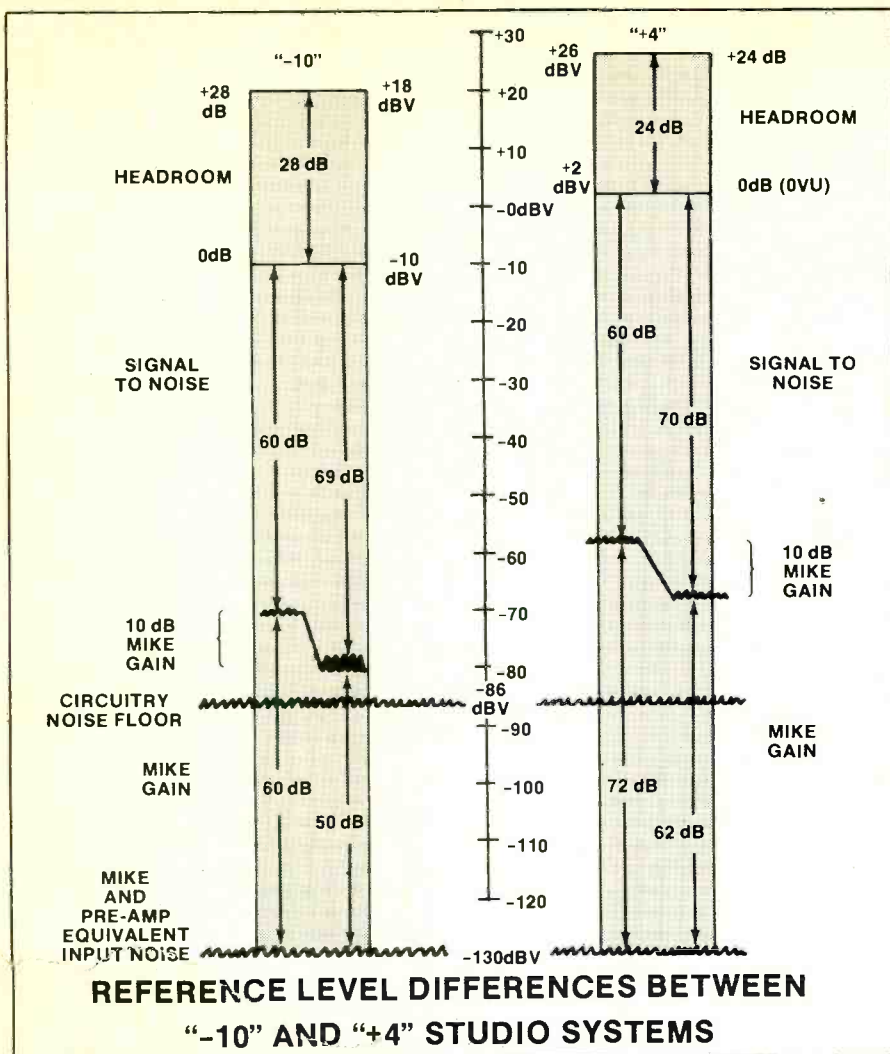
REFERENCE LEVEL DIFFERENCES BETWEEN "-10" AND "+4" STUDIO SYSTEMS

"-10"

+28 dB	+18 dBV
HEADROOM 28 dB	
0 dB	-10 dBV
SIGNAL TO NOISE 60 dB	69 dB
10 dB MIKE GAIN	
CIRCUITRY NOISE FLOOR	-86 dBV
MIKE GAIN 60 dB	50 dB
MIKE AND PRE-AMP EQUIVALENT INPUT NOISE	-130 dBV

(dBV scale: +30, +26 dBV, +24 dB, +20, +10, -0 dBV, -10, -20, -30, -40, -50, -60, -70, -80, -86 dBV, -90, -100, -110, -120, -130 dBV)

"+4"

+26 dBV	+24 dB
HEADROOM 24 dB	
+2 dBV	0dB (0VU)
SIGNAL TO NOISE 60 dB	70 dB
10 dB MIKE GAIN	
	MIKE GAIN
MIKE GAIN 72 dB	62 dB

---

volts. If these voltages are substituted into Equation 3, we get:

$$n = 20 \log (1.228/0.316)$$
$$n = 11.78 \text{ dB}$$

Thus the +4 gear has an almost 12 dB hotter 0 dB level than the -10 system.

Now onto the promised comparison, as outlined in the accompanying diagram.

First, the Theoretical

To facilitate this comparison, I have set up two bar graphs with an absolute dBV (1V) indicia running between the two; each system's relative dB ratios are shown to either side. I have had to make a number of assumptions, and these graphs are not representative of any one manufacturer's console, but what I feel a competent designer would come up with given the same constraints.

I assume identical high-quality microphone pre-amps with -130 dBV EIN (equivalent input noise). (Note: the +4 system's signal receives an additional 12 dB of gain to bring levels up to 0 dB.) I further assume that both consoles use low-noise integrated circuits (BIFET or equivalent), and lastly that the -10 system uses a single-ended output with ±15-volt power supplies, versus the +4 system's differential output and ±22-volt power supplies.

The results of such a +4/-10 comparison are a little surprising. The signal-to-noise ratios of both systems are identical, and dominated by the microphone's thermal noise; the -10 board actually delivers a 4

dB better headroom spec! We would have to assume a 10 dB hotter signal (10 dB less microphone gain) before the circuitry noise floor degrades the -10 system's SNR by even 1 dB, and that's assuming what I believe to be conservative figures for circuitry noise floor.

Really Now?
(The Practical Comparison)

While the theoretical comparison is an interesting exercise, it only shows what can be done. Most studio equipment designed to operate at -10 dB levels is engineered to be lower cost, which often results in less than SOTA microphone preamps, and noisier electronics. Another factor that doesn't show up in the theoretical comparison is non-thermal noises, such as hum, RF, etc. I've yet to see a studio that didn't have any; the diligent ones keep it under control, but it's always there somewhere.

The +4 studio has the advantge of almost 12 dB more signal (15 times the power) in line-level feeds. Broadcasters use a 0 dB system level that's even another 4 dB hotter: +8 dBm (600).

In conclusion I must give the +4 studio a slight edge in ease of obtaining a clean signal, but by no means is the -10 studio out of the running. Properly set up and properly used, the available dynamic range of both is likely to be limited by other factors. Ho hum, another wishy-washy answer.

Reading for Extra Credit:

My first reference is simply the IEEE Standard Dictionary that I have used for numerous definitions. Since this reference is dated 1977, there may be changes already in the works, and I would appreciate hearing of any more recent references that contradict Std 100-1977.

Reference #2 is a detailed specification defining True VU meters.

References 3 and 4 are articles containing general information about noise and console specs/performance.

1. IEEE Std 100-1977; "IEEE Standard Dictionary of Electrical and Electronic Terms" (second edition).

2. USA C-16.5-1961.

3. "Fact or Fiction: Console Noise Specifications," by Paul C. Buff; R-e/p, December 1977.

4. "Performance Limits in Contemporary Console Design," by John Roberts; R-e/p, April 1980.

PS — As I have only recently arrived at this heightened awareness of proper usage, don't be surprised to find some of the "common" mistakes show up in my past writing.

PSS — (Opinion) As a circuit designer I find the use of the decibel irresistible for describing voltage gains (yes, even transformers), usually with no pretense of power ratio equivalence between input and output circuit. (How many power amps do you know of with 8-ohms input impedance?) I accept that this may be a theoretical no-no, but until a voltage ratio-only equivalent for the decibel arrives, I see no alternative. ▪▪▪

news

QUAD-EIGHT BECOMES QUAD EIGHT/WESTREX

Quad-Eight Electronics has purchased from Litton Industries its Westrex Sound Recording operation in the U.S., and all Westrex operations in the United Kingdom. The merged business will operate as Quad Eight/Westrex.

Westrex, originally the motion-picture sound equipment division of Western Electric, was instrumental in the development of sound for motion pictures over 55 years ago. The Westrex operations were acquired by Litton in 1958.

Although Quad-Eight has always sold at least 50% of its products internationally, as Quad Eight/Westrex it will now have a multinational manufacturing base with the acquisition of Westrex's manufacturing facility in the U.K. The company will manufacture both recording equipment and consoles in the U.S. and U.K., allowing a greater world-wide distribution system for all of its products.

dbx TO OFFER CPDM CIRCUIT CARDS ON OEM BASIS; DESIGNING CPDM-TO-PCM TRANSCODER

The Companded Predictive Delta Modulation digital recording technology, originally developed for the dbx Model 700 Digital Audio Processor, will be made available to manufacturers of tape
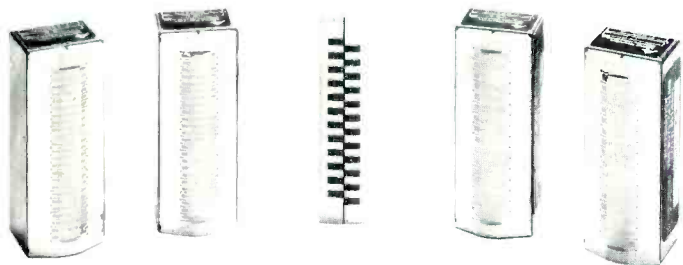
24 & 16 TRK
Replacement Tape Heads
For MCI Recorders

¼" & ½," 2 TRK
Replacement Tape Heads
For MCI Recorders
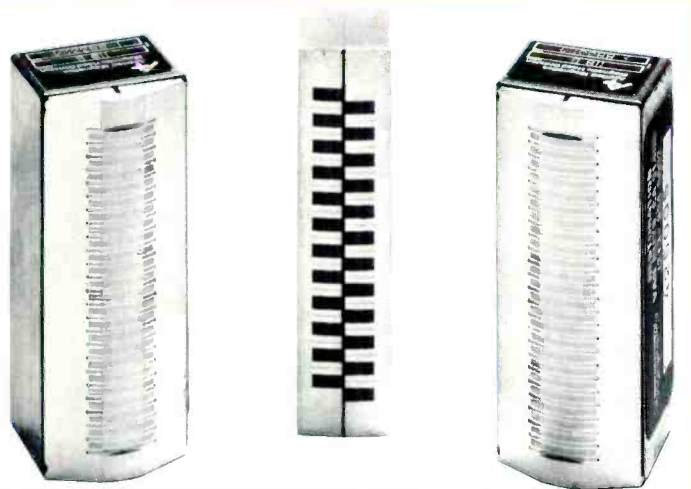
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TRACK CONFIGURATIONS

CASSETTE

1 CHANNEL 2 TRACK
RECORD/PLAY
5100 SERIES (W1R/W1P)

.150"

.056
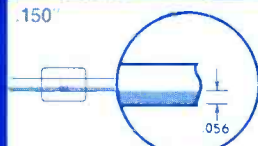
2 CHANNEL 2 TRACK
RECORD/PLAY   ERASE
5200 SERIES (W2R/W2P) 5210 SERIES (W2ER)

.150"

.092 ℄ TO ℄

.057
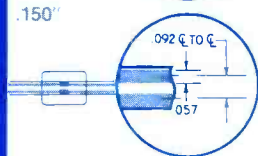
2 CHANNEL 4 TRACK
RECORD/PLAY
5230 SERIES

.150"

.035 ℄ TO ℄

.021
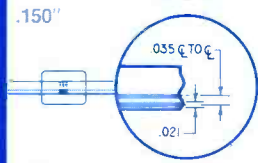
4 CHANNEL 4 TRACK
RECORD/PLAY
5410 SERIES (PW4J)·

.150"

.021

.054 ℄ TO ℄

.035 ℄ TO ℄

· AVAILABLE IN DURACORE·

8 TRACK

2 CHANNEL 8 TRACK
RECORD/PLAY
5810 SERIES (P-B2L)
5800 SERIES (B2L)·

1/4"

.127 ℄ TO ℄

.021
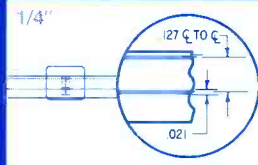
4 CHANNEL 8 TRACK
RECORD/PLAY
5830 SERIES (P-BQL)

1/4"

.0635 ℄ TO ℄

.021

OPEN REEL

FULL TRACK
RECORD/PLAY   ERASE
9100 SERIES (PR-B1F)·   9125 SERIES (PR-B1EF)
                8100 SERIES (B1EF)
                4400 SERIES (SMEF)
                4400L SERIES (LMEF)

1/4"

.240 TRACK
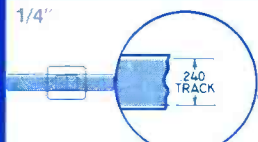
2 CHANNEL 2 TRACK
RECORD/PLAY   ERASE
9200 SERIES (PR-B2H)·   9221 SERIES (PR-B2EH)
8200 SERIES (WP-B2H)   8300 SERIES (B2EH)
2000 SERIES (P-B2H)·   2200 SERIES (SSEH)
                2200L SERIES (LSEH)

1/4"

.156 ℄ TO ℄

.080
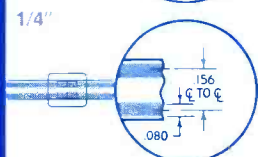
2 CHANNEL 4 TRACK
PLAY/RECORD   ERASE
8600 SERIES (WP-B2Q) 8700 SERIES (B2EQ)
1200 SERIES (P-B2Q)
1000 SERIES (B2Q)

1/4"

.043

.136 ℄ TO ℄

·AVAILABLE IN DURACORE·

STUDIO SERIES

3 CHANNEL 3 TRACK
RECORD/PLAY
5700 SERIES (P-B3Q)·

1/4"

.043

.100 ℄ TO ℄

4 CHANNEL 4 TRACK
RECORD/PLAY
5600 SERIES (P-BQQ)·

1/4"

.037

.071 ℄ TO ℄

4 CHANNEL 4 TRACK
RECORD
9400 SERIES (STR-4B)
PLAY
9400 SERIES (STP-4B)
ERASE
9400 SERIES (STE-4B)

1/2"

.070

.130 ℄ TO ℄

8 CHANNEL 8 TRACK
RECORD
9800 SERIES (STR-8B)
PLAY
9800 SERIES STP-8B)
ERASE
9800 SERIES (STE-8B)

1"

.070

.130 ℄ TO ℄

16 CHANNEL 16 TRACK
RECORD
9990 SERIES (STR-16B)
PLAY
9990 SERIES (STP-16B)
ERASE
9990 SERIES (STE-16B)

2"

.070

.127 ℄ TO ℄

24 CHANNEL 24 TRACK
RECORD
9983 SERIES (STR-24U)
PLAY
9984 SERIES (STP-24U)
ERASE
9987 SERIES (STE-24U)

2"

.037

.084 ℄ TO ℄

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Take control of your drum sound with ATM microphones by Audio-Technica. We have all the right models to deliver the sound that moves your group and your audience. Our ATM11R, for instance,

goes down to 30 Hz and is crisp and smooth all the way up to 20 kHz. What you play is *exactly* what they hear! Move in as close as you wish without distortion or overload. The ATM11R can mike *anything* you play except the kick drum (and we've got a killer mike for that)!

The great sound and high headroom (141 dB SPL) of the ATM11R comes from our advanced Fixed-Charge electret element, remote-powered from your mixer. It has an ultra-thin diaphragm (just *4 microns* thick) to respond instantly and accurately to every sound you make.

And don't worry if you accidentally hit an ATM microphone. They're all-metal and really Road Tough. No fragile plastic to shatter like so many others. And

very small to keep your kit clean, uncluttered, and easy to play.

Now about the kick drum... the ATM21 Unidirectional Dynamic will give you the crisp, powerful beat you're looking for. No distortion, even when buried in your hottest drum. In fact, it's a great all-around dynamic for close-miking your entire kit when remote power isn't available.

To find out how ATM microphones can help your sound, visit your nearest Audio-Technica dealer. Or write for our colorful catalog. We've got everything you need to make your drum sound the very best part of the mix!

Model ATM11R Unidirectional Remote-Powered Condenser Microphone. Under $160.

audio-technica
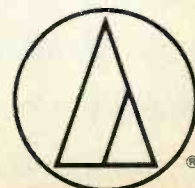
"Because of this a lot of eight-track studios began to pop up in England. This was coincidentally all taking place with the decline of the 'Megabucks' in the recording industry. At the same time record budgets were plunging, this underground thing was happening where people were saying we don't need all that technology to record. This was all perculating through my mind when I realized that with my experience as an engineer, and with the right mikes and so on, I could probably get an excellent sound by building my own studio. It's just a matter of more pre-production planning.

"After seeing all of this rawness coming out of Europe, and the critical acclaim that was going to some of the garage albums that LA bands were cutting, I decided I could combine elements into a high-tech meets low-tech recording philosophy."

The best example of this philosophy can be found in DeeJay Studios, the label's recording facility, and which represents something of an inventive approach to the music industry. "I have an Otari MX-5050 MkIII eight-track — it's totally state-of-the-art, and has all of the features that most 24-tracks have. It also has a signal-to-noise ratio and frequency response that's as good as most 24-tracks.

"The important thing about the recorder for the way I use it is that it is SMPTE compatible. This allows us to cut our basic tracks in my studio, transfer them up to 24-tracks, and then the 24-track becomes the master. We stripe the 24- and eight-track tapes with SMPTE, and then make slave copies. We make several slave tapes of each song by putting the timecode on track #8, and then a rough mix of the basic track in stereo, giving us five tracks to play with. We may use one slave tape for the vocals, one for the percussion, and another for additional guitars. We then lock those up to the 24-track and transfer them up.

"Everything is one generation down, but that loss is offset by the fact that our 24-track master has made only a few passes, giving it better sound quality than a normal master that has been shuttling back and forth for the entire recording. When you're working with a rock 'n' roll band you can spend a long time tracking if you want to do it right, and this process gives me the freedom to do that. I wouldn't have that freedom if I was paying for time in a major studio."

Recording this way, according to Degher, is bringing back experimentation into the studio. "We can take more time to experiment and work on sounds than a lot of acts with major recording budgets in a big studio, allowing us to get a sound quality on a par with major studios. In the end we have a 24-track master that can be mixed in any of the major studios, giving us a totally high-tech sound.

"My console is very clean and very punchy because that's what I need. It was custom built by Dan Kipman and it has a lot of Trident A-Range microphone preamps, as well as some Jensens, so we're getting the same kind of signal path you find on a really expensive console. We kept every gizmo to a minimum on each module so that the signal is not processed at all, giving us maximum flexibility when we mix in a major facility."

There are a few things one has to be aware of when recording this way as Degher points out. "You really do have to be aware of the proper way to use SMPTE timecode. You should always put the SMPTE on the highest track, and it's a good idea to leave an open track next to it. SMPTE sounds like it's easy, but in fact it is very high technology. If you do it incorrectly you'll end up with nothing because the tapes won't lock back up. You can't just go in and throw a timecode on there and transfer it across. There's a lot of steps you have to know.

"Obviously you have to work with a major studio that has the SMPTE gear. You have to be prepared to pay for time in that studio when you're doing your SMPTE work, and your mixing. You should be able to transfer your basics and do your rough mixes to make your slave tapes in one session, so it's not that expensive.

"Another advantage is that we're already geared for video because of having the SMPTE code on the master 24-track tape. People spend a fortune on video without getting good sync, we're almost assured of that."

DeeJay Records is looking for new signings, according to Degher. "Now that Darius & The Magnets is to the point where other people are active in their career, I am looking at other acts to work with after the new album is finished. John Collins of Golden Image Management is managing Darius & The Magnets, and his firm, New Image Public Relations, is doing their press, so I will have the time to work with another artist once the album is done.

I've had to become a legal beagle and launch publishing ventures and so on to give DeeJay Records an adequate base, but now we're ready to expand."

With Degher's attitude toward making technology work for instead of against him, and his "Create Your Own Destiny" philosophy, he has clearly shown that being inventive in these times is not only a necessity, but a key to the future as well.

■■■

news

machines on an OEM basis, in the form of circuit cards to be interfaced with tape transports to produce open-reel, fixed-head CPDM digital recorders. Such a move will enable recorder manufacturers to offer CPDM machines that will be compatible with each other, unlike the various PCM recorders on the market that currently are not all compatible with one another. To further ensure the compatibility of all recorders equipped with dbx digital (CPDM), dbx is also specifying the track format to be used.

The CPDM-to-PCM transcoder will accept CPDM bits as an input from any manufacturer's two-track machines equipped with dbx digital. The transcoder will then convert, in the digital domain, CPDM bits into PCM bits in a format suitable for mastering of Compact Discs.

POTENTIAL PITFALLS IN MASTERING FOR THE COMPACT DISC

by Roger Nichols

The purpose of this article is to clear up a few misconceptions relating to the quality of digital audio. Hopefully it will be worth the time it took to write it.

It looks as though we are in the middle of a technological "bit war" — Analog versus Digital. The most widely publicized battle at present seems to be over Compact Discs. A lot of people, some quite prominent in the music business, are attempting to "byte" off more than they can chew.

There is so much ammunition flying around that it is hard to find a place to start. I'll just put on my flack jacket and high boots, and jump in the middle.

I have noticed quite a few reviews of the Compact Disc format comparing a digital release to its analog counterpart, and denouncing the Compact Disc as being inferior to the vinyl version. Comparisons like this are good, and should continue. It is a shame though, that after enjoying a photograph of an apple for years, your first real apple is infested with worms.

To compare an analog pressing and a Compact Disc of the same product, they should both be produced properly — that is, from the *correct* source material. Just to clarify the process, I will use the following example (ficticious of course):

Artist: Doo Wah Joel.
Album: Nylon Stockings.
Producer: Phil Producer.
Multitrack: 3M 32-track Digital Mastering System.
Mix: Sony PCM-1610 digital two-track processor and U-Matic VCR.

So, up to this point we have a well recorded, completely digital album. The tape used to produce the lacquer masters for the analog pressings was the original digital two-track VCR master. The analog version seems to be as good as it can possibly be.

During the mastering process, an equalized, Dolby-encoded, analog 15 IPS tape copy was produced per the record company's instructions. Why? "Because we always order an equalized tape in case we need to make copies." I guess if someone needs a cassette . . . So far, so what?

Now someone decides that this album should be made available in the new Compact Disc format. The Compact Disc

department calls down to the production department, and orders a Sony PCM-1610 copy of the album for the Compact Disc pressing plant. The production department whips out a "production master" of the album, and transfers via the Sony PCM-1610 to a U-Matic vedeocassette. (They didn't want to wear out the equalized tape, so they made a copy of it and called it the production master; this is the tape they *always* use to produce requested copies.) Ta da! We now have our Sony PCM-1610 tape with which we will produce our Compact Discs.

So, the plot thickens! Now we have an analog pressing produced from the original digital two-track master, and we have a digital pressing produced from a second-generation 15 IPS analog tape copy!

Give me strength! If the analog pressing didn't sound better than the Compact Disc in this case, I'd eat my F1.

A digital recording will not be any better than the source material — it is not supposed to! If you make a Compact Disk from an inferior tape, you are going to have an inferior Compact Disc. The Compact Disc *must* be made from the best possible source: the *original* two-track master.

I tried an experiment. Having recorded a live source onto a Webster Corp. wire recorder, I then transferred the recording to a Mitsubishi X-80 digital two-track recorder. When I played them both back I couldn't hear *any* difference at all. Does this mean that the $25,000 digital machine

— **Roger Nichols** —

— **the author** —
Roger Nichols has been a recording engineer for the past 15 years. He has engineered all of the albums by Steely Dan, which have won him three Grammies and five Grammy nominations. His credits also include the Donald Fagen Nightfly album, which was recorded and mixed digitally; he also recorded and mixed the last two John Denver albums to digital. Nichols has used the 3M, Sony, and Mitsubishi digital multitracks, as well as the Sony, JVC, Soundstream, 3M and Mitsubishi two-tracks. In the fall of 1982 he recorded 28 John Denver concerts in Europe using a pair of Sony PCM-F1 processors synchronized to provide four-track capability.

For additional information circle #19

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is no better than the wire recorder I picked up for 50 cents at a garage sale? No, but it does mean that the digital tape copy is *no better* than the recording made on the wire; it is an *exact* copy.

Who's to Blame?

The problem with Compact Discs is not the medium, but the record companies and the uninformed people shoveling the product out into the marketplace without paying attention to what they are doing.

The consumer is not as stupid as the record companies make him out to be. If the Compact Disc falls flat on its face, it will not be the fault of the digital medium, but the fault of the record companies who insist that "Nobody will know the difference, let's just do it the easiest and quickest way we can." This is surely a penny-wise/dollar-foolish attitude.

The consumer will *not* accept an inferior product; they are already screaming about Compact Disc quality (or lack thereof) in Europe and Japan.

Now that I have done some screaming of my own, I would like to add some constructive comments.

Not all of the record companies are trying to do everything wrong. I have personally followed some Compact Discs through the entire process — from recording, mixing, mastering, CD master, to Compact Disc — and compared the disk with the original two-track digital master. When done properly, the results are *amazing*.

If anyone wants to compare an analog pressing with the Compact Disc version done the right way, then run out and get a U.S. copy of the Donald Fagen *Nightfly* album, and tell me which one sounds better. (Please see the note below on which "version" of the Donald Fagen album to listen to.)

If you get a Compact Disc that sounds bad, complain! Not about the digital medium, but to the record company that produced it. Demand they do it over again — the right way. And make sure they replace your bad disk with a good one when they fix it. It's time for the record companies to treat Compact Discs as records, and to master them the way that records are supposed to be mastered.

For those people that are too embarrassed to ask how to correctly produce a Compact Disc master, included at the end of this article is a guideline of the proper procedure. If any of the record companies wish further assistance, I am more than willing to help.

Please! I am running out of space to store my bad Compact Discs.

One quick note on the *Nightfly* album. As with any manufacturing process, there are sometimes mechanical problems not caught in time. If you plan to make the comparison mentioned above, make sure that you get the domestic U.S. Compact Disc release of the album. There are two ways to tell which one you are getting. First, the U.S. release is packaged in a 6- by 12-inch box, with information about the disk and the artist printed on the outside of the box. The imported versions are just the Compact Disc in its 5- by 6-inch plastic case, with no outer packaging. The second way to tell the good one from the bad one is to look at the laser imprinted number located on the music side of the disk near the center hole; the number consists of the album number and some production information. The number of importance is at the end of the sequence in the clockwise direction. The bad disk number ends "021 02," while the good disk number ends 021 03."

PROCEDURE FOR PRODUCTION OF COMPACT DISC MASTER TAPES

For Albums Mixed Digitally:

1. Contact the producer of the project. Tell him that you require a Compact Disc master tape. Let him be involved.

2. Locate and use the *original* digital master. *Not* the EQ copy!

3. Arrange to master the Compact Disc at the facility that originally mastered the analog release. (Note: there is a reason that the producer and artist chose this mastering facility; those reasons still apply to Compact Disc mastering.)

4. Tell the mastering facility that you will be mastering for Compact Disc, and that you will need the equalized tape to be in the following format: Sony PCM-1610, JVC DAS-90, or Mitsubishi X-80. The format needed will depend on the pressing facility being used (i.e. Denon, Toshiba, Sony, Polygram, etc).

5. Tell the mastering facility in which format the original two-track digital master was recorded, so that they can obtain the proper playback machine (i.e. 3M, Sony PCM-1610, JVC DAS-90, Mitsubishi X-80, Soundstream, etc).

6. It is up to the producer to decide whether the same EQ and level adjustments as made on the analog master are to be used during Compact Disc mastering. (For example, if minimal EQ and level changes were used, such as 1 dB boost at 12 kHz on one tune, and a 1 dB boost on another, then in all probability these settings will be fine for the Compact Disk. If heavy high-frequency limiting or significant level drops are required for a tune placed in the inner diameter of the analog master, then these settings will need to be changed for the Compact Disc master, since there are no such restrictions with digital.)

7. The tape you receive from the mastering facility is an equalized digital master. Because sides of an analog album are cut separately, the equalized digital master must be edited so that there is no break between sides one and two.

8. The edited tape now becomes the Compact Disc master. The method of arriving at the edited version of the tape depends on the tape format required by the pressing facility:

(a) If the format is to be Mitsubishi X-80, then the equalized digital master is edited with a razor blade, and the equalized digital master then becomes the Compact Disc master.

(b) If the format is to be JVC DAS-90 or Sony PCM-1610, then the equalized digital master must be electronically edited, a method similar to videotape editing. The equalized digital master remains intact, and the digitally edited copy becomes the Compact Disc master. Because it's a digital-to-digital process, there is no generation loss.

In both cases, SMPTE timecode must be

added to the Compact Disc master; the timecode must be locked to the sampling frequency.

9. The timecode for the start and end of each selection on the album must be accurately noted to the frame. Each pressing facility requires that a different offset value be added or subtracted from these figures to produce the timecode number that is entered onto the Compact Disc order form. Follow these directions exactly.

10. Make a Sony PCM-F1 digital copy of the Compact Disc master. *No analog copies!* You can't okay a digital disk from an analog tape copy. Send the F1 copy to the producer for approval. This is the same procedure as approving a master ref or test pressing of an analog record.

There have been instances where a few seconds of an intro to a song were cut off accidently during the digital editing process. It is much easier to catch these errors at this stage than when the Compact Discs are in the stores. Not all producers have an F1 yet. Warner Bros. and Motown have extra sets to loan to the producers so that they can approve the Compact Disc masters.

11. From the Compact Disc master, two digital copies must be produced with regenerated timecode. This is easy to do. There is no generation loss.

12. One of the digital copies must be listened to from beginning to end to make sure it is perfect. This copy is labeled the "A" copy, the other copy the "B" copy.

13. Both the "A" and "B" copies are sent along with the timecode list to the Compact Disc pressing plant.

14. The Compact Disc master is retained for future use and archiving.

Albums Mixed Analog:

All procedures are exactly the same except for the following amended steps:

2. Use the *original* two-track *master!* Do *not* use the equalized copy! As sacred as you think the original two-track master is, that's the tape you must use. The people handling the tape are professionals who know how to take care of it. Besides, the digital equalized master you end up with will not be a generation down, and in 10 years it is going to be in much better shape than the analog tape stored in a Bekins warehouse.

5. Telling the mastering facility the format of the original is not as important with analog masters. The tape will usually be either half-inch/two-track or ¼-inch/two-track. They can handle it.

14. Remember, this new digital master can be used to archive your precious album.

Old Catalogue Albums:

Follow the same procedure as for Albums Mixed Analog, except as follows:

1. Sometimes the original producer and artist are not available to supervise the project. In this case, involve someone who can make the proper judgements regarding the mastering of this album. A good choice would be the mastering engineer, or an available engineer or producer working on another Compact Disc project.

2. Use the *original* two-track *master!* This time it is even more important that you use the right tape — get a digital master of this project before the original master turns to dust in your hands!

3. If the original mastering facility is not available, pick one of the best places around. They know what they are doing. This album has already paid for itself; the Compact Disc sales will be all gravy. Get somebody who will make it sound good. The rewards will more than make up for the extra five dollars it costs to do it right.

Compilations — "Best of Albums":

This procedure requires a couple of extra steps — important ones:

2. Use the original two-track masters of each of the albums being compiled, whether they are digital or analog. Do *not* use an *EQ* copy.

2.5 Align the master playback machine to the master tape, and transfer the required material to a digital tape machine. Repeat this process for each master tape until all the material to be compiled has been transferred. You now have a compiled master without a generation loss.

Continue with step #3.

The reason why the tunes must be transferred from the original masters is that different albums are usually done at different studios, on different tape machines, using different types of tape, with different alignments. Therefore, if a playback tape machine is aligned to properly reproduce one master tape, it will not playback another master correctly. The playback machine *must* be re-aligned for each subsequent master tape. Since you must play all of the tunes on one side of an album without stopping when cutting the master disk, there isn't enough time to change tapes and re-align machines between selections. ∎∎∎

FIRST IMPRESSIONS — An In-use Equipment Assessment
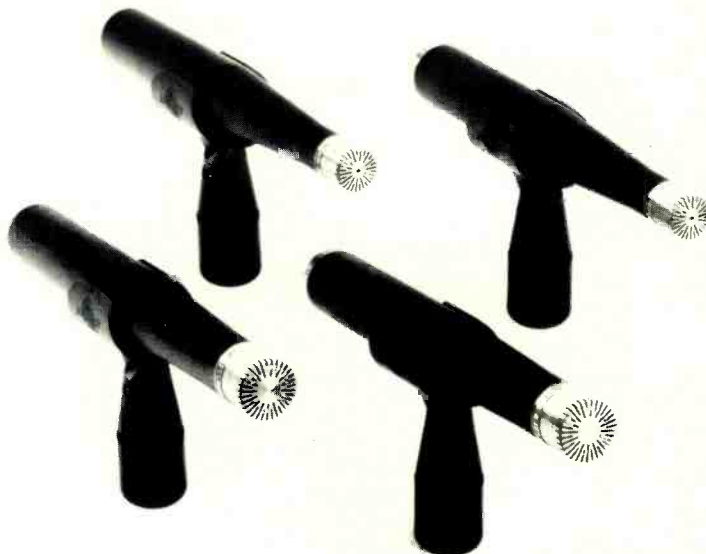
BRUEL & KJAER 4000 SERIES CONDENSER OMNIDIRECTIONAL STUDIO MICROPHONES

The recently introduced 4000 Series condenser microphones from Bruel & Kjaer are noteworthy in several respects. In keeping with B&K's established range of instrumentation and calibration mikes — which, interestingly, despite their intended measurement application are used by a number of session engineers to record a wide variety of instruments — all four models that make up the new series feature omnidirectional polar patterns. And, while the majority of engineers are possibly more used to working with dynamic and condenser cardioid or directional mikes — to ensure adequate acoustic separation during multimike/multitrack sessions, or as a stereo coincident array — B&K makes a good case for offering omnis to the studio recording industry. In its product literature, the company makes reference to the following advantages: freedom from fringe effects, owing to a broad amplitude response extending well beyond the audio range; smooth phase response; high on-axis/off-axis uniformity for a clean, transparent, and well-balanced sound; wide usable dynamic range; low sensitivity to boom or stand movement, and wind/breath induced noise.

And, commenting on the cardioid-versus-omnidirectional argument, and associated sound-isolation problems, B&K offers that in those situations where a cardioid might seem more appropriate to ensure maximum sound separation, the amount of leakage can also be controlled by altering the source-to-microphone distance. Since an omni microphone does not exhibit any proximity effects — in particular, an increase in bass frequencies — it can be placed closer to the instrument to reduce sound leakage from nearby sources.

The 4000 Series currently comprises four models: 4003, 4004, 4006, and 4007. The 4003 and 4006 are acoustically identical mikes with a quoted on-axis frequency response of 20 Hz to 20 kHz, ±2 dB, and very low equivalent noise levels of typically 15 dBA. Types 4004 and 4007 feature a quoted on-axis response of 20 Hz to 40 kHz, ±2 dB, and can handle peak sound-pressure levels of 168 dB before clipping. Each basic design is available as a Line-level version — 4003 and 4004 — for use with B&K's Model 2812 130-volt combined power-supply and pre-amplifier unit, or as a standard P48 Phantom version — 4006 and 4007. Use of the Model 2812 line-level PSU/pre-amp unit is said to provide a higher output level (in balanced mode, 18 dB higher than the corresponding phantom-powered versions, for increased headroom) and, by dispensing with coupling transformers, to avoid core saturation at low frequencies, and improve amplitude, phase, and distortion performance.

FIRST IMPRESSIONS
B&K 4000 Series Omnis

To discover more about the potential applications of 4000 Series microphones in "typical" recording environments, R-e/p arranged for operational assessments with two leading engineers: **Bernie Kirsh**, and **Shawn Murphy**. While, because of time limitations, we were unable to try out the B&K condenser microphones on as wide a range of instruments and material as we would have liked, the results from this necessarily brief trial were certainly impressive.

Bernie Kirsh is chief engineer of Chick Corea's Mad Hatter studio in east Los Angeles. Various combinations of 4000 Series mikes were used on a selection of instruments, and compared with conventional studio microphones during basic tracking dates for a jazz ensemble session, comprising drums, bass, acoustic piano, and acoustic guitar. A Model 4004 (high-level/130-volt PSU) placed on the kick drum produced a "more mellow and fuller body" sound than Kirsh's normal choice, even when mounted very close to the rear skin near the beater. On piano, a 4003 (low-noise/130-volt PSU) on the bottom strings, and a 4007 (high-level/48-volt phantom) on the high strings produced a "balanced, open sound," and one that compared very favorably with conventional studio condenser mikes.

On acoustic bass a 4003 positioned about 1 foot from the F-hole resulted in a "more open and 'airy' sound," Kirsh comments, "with slightly more top-end and string noise. And, by moving the mike closer in towards the bass, the sound increases in body and warmth. All in all,

the 4003 is very quiet in operation, and has an added 'presence' over other mikes."

For acoustic guitar, Kirsh elected to try another 4003 mounted about six inches away from the fingerboard, and pointing at the bridge and sound hole. "The 4003 reproduces the sound of an acoustic faithfully," he offers, "and picks up more 'bloom' than other mikes. Also, as the sound is moving through and away from the instrument, the B&K is still 'hearing' it. It is also capable of capturing the percussive transients produced."

One comment that Kirsh had to offer was that the sound produced by the B&K was a "little less exciting" than that from his normal choice of microphone set to omnidirectional pattern. He was quick to concede, however, that certain kinds of added coloration produced by conventional mikes might be the type of sound studio engineers are more used to, and that they might have to adjust their assessment in light of the more faithful and accurate sound produced by a B&K model.

Kirsh did acknowledge one potential problem that might be encountered when using B&K omnis during a multimike ensemble session — sound leakage. In particular, because of the 4000 Series' extended high-frequency response, the sound from instuments such as cymbals and high-toms can easily spill into nearby mikes. That being the case — and it was certainly true that crash and ride cymbals leaked into the kick and piano mikes — he offers that the models under assessment might be more appropriate for two- or three-mike spaced-omni distance miking of jazz and orchestral sessions, or for overdubs of single acoustic instruments. On acoustic-guitar or piano overdubs, he offers, the "beautiful high degree of sound definition, and

top-end response" makes the B&K 4003 an ideal choice.

Shawn Murphy is a Los Angeles-based music scoring mixer who works primarily on a contract basis for Walt Disney Productions, where he recorded and mixed the digital re-issue soundtrack of *Fantasia* (see R-e/p October 1982 issue). In addition, Murphy has recorded most of the music for the many EPCOT Center film presentations. Recent work outside of Disney includes music soundtracks for *Brainstorm* and *Testament*.

Murphy made extensive use of the B&K microphones over a period of two weeks, with direct comparisons being made to other high-quality condenser omnidirectional models.

"First and foremost," he notes, "they really are good-sounding mikes. The mikes sounded clearer and more revealing than anything I compared them to."

The clarity became evident during percussion overdubs, when Murphy used a 4007 (high-level/48-volt phantom) omni. "The mike had good clarity, good impact, and good low-end. In addition, it was not fussy in terms of positioning. I heard more of the instrument; it was more like I was in the room [than it would be with his normal mike choice]."

Murphy noticed the same "you-are-there" presence of the B&K mikes during a narration recording session. However, he became quickly aware that a function of the good design was that the microphones "tell you how good or bad your acoustical situation is. If you have someone reading narration from a music stand, you can hear the bounce off the stand. You can hear whatever baffling you have around the person, and you can hear the room *very* clearly.

"So, if your narration booth has a slap off the

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glass, or if the situation has any air conditioning rumble or other noise that you don't normally hear, these mikes tend to reveal them."

Such a lack of coloration would preclude, Murphy says, his using the B&K omnis in all but the best situations, although he is quick to note that he "would love to hear them in a concert-hall situation with known good acoustics. Around here, maybe Royce Hall [UCLA] or the Ambassador [Auditorium in Pasadena, CA]. I'd even like to hear them in MGM Stage One, which is a terrific-sounding room."

Along the same lines, Murphy offers that he uses coincident pair and MS stereo microphone techniques less than he otherwise might because of *their* revealing nature.

"I find that [in an average recording studio] I use two-thirds overall and one-third spot mikes. When I use an MS mike, I use two-thirds spot mikes to get added definition.

"Omnis . . . would be more useful for me than an MS [mike] due to the flexibility of positioning three [spaced] mikes versus one. Also, with an omni you are going to get a better low-end and, theoretically, a better high-end than the MS configuration."

During one scoring session at Walt Disney Studios, Murphy compared a pair of 4003 (low-noise/130-volt PSU) mikes mounted next to his left and right M-50 mikes. "In this room I normally use M-50s instead of the [normal omnis] because I find that even with [my normal choice], the room reflections are not good enough sounding to where you would want to let an omni operate as an omni.

"The M-50 has a phenomenal low-end and a directional high-end, which allows you to control the perspective and the distance from the orchestra that you are miking, so that if you have a very large orchestra you can back it off. And if you want a little more definition in one section, you can just move the mike."

"Compared to [my normal omnis], the B&K has less 'personality.' The high-end [of my omnis] is a little more forgiving, and I think it narrows a bit on the high end."

Murphy thought that the low-noise/130-volt power-supply version was "noticeably quieter, and had a different-sounding high-end than the phantom version. I wonder if it was caused by difference in capsules, or in the powering scheme.

"Of course, it could be that because the trans-fomerless power-supply version gives you a very high output it comes out basically, or nearly, at line level, and I was able to pad the console input down far enough so that the [console] pre-amp noise was less of a factor.

"In any event, gitting rid of the transformer means that, theoretically, you get better low-end and better transient response and, theoretically, a better-sounding mike."

The only problem Murphy forsees with the 130-volt powered microphones (4003 and 4004) is a potential high-end loss with a long cable from the mike to the power supply.

"If you have a RF problem, or any kind of interference, you should be careful. Once it gets to the power supply it's great — the output is so hot that anything getting to the console is no problem. The only time you might have a problem is in a remote situation, such as that in a concert hall, where you might have to fly the mikes up where there is no AC power." □□□

news

LEXICON RECIPIENT OF MONITOR AWARD FOR ENGINEERING ACHIEVEMENT

The 1983 Video Production Association's Monitor Award for Engineering Achievement has been presented to Lexicon for introduction of the Model 1200 Time Compressor, and in recognition of the company's contribution in the area of digital processing.

"The Model 1200 Time Compressor has allowed variable speed film to tape and tape to tape transfers while maintaining a high standard of audio quality," according to the VPA's Monitor Committee. "Altering program segment lengths during the editorial process is a service that producers have come to expect. The Model 1200 is the most widely requested device to this end."

Shown above with the award are Ronald P. Noonan, president, and Charles Bagnaschi, vice president and director of engineering.

SPARS/UNIVERSITY OF MIAMI DIGITAL AUDIO SEMINAR

The three-day seminar, entitled "The Digital Revolution: In Search of a Peace Treaty," to be held at the University of Miami, March 8 thru 10, 1984, is intended to define the proper role to be played by digital technology — both today, and in the future — and to shed some objective light on the "state-of-the-digital art."

Seminar I, **An Introduction to Digital Audio**, to be given by Ken Pohlmann, director, Music Engineering Technology, University of Miami, is designed as a review of the basics of digital technology. In addition, it will offer participants a brief status report on digital audio today.

Siminar II, **Digital Audio On Trial: If It's So Good, Why is it So Bad?**, will be moderated by Michael Tapes of Sound Workshop; For the Prosecution: Doug Sax of The Mastering Lab; For the Defense: a speaker to be announced; with Witnesses John Eargle of JBL, and Len Feldman.

Digital audio has been widely praised by its promoters, and often damned by the critics. Are the promoters trying to sell the recording industry an immature technology? Are the critics expressing nothing more than the usual distrust of new technology? Is there something inherently wrong with the concept of Pulse-coded music?

Seminar III, **Digital in Perspective**, will discuss the proper role of digital technology. Is it supposed to (eventually) replace everything else, or are there some

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tasks that will always be better left to analog? At this time, are we making unrealistic demands on the medium?

Seminar IV, **Digital for Dollars**, will be moderated by Hamilton Brosius of Audiotechniques, with panelists Bruce Botnick of Digital Magnetics, and Joe Tarsia of Sigma Sound.

Seminar V, **Review of the Proceedings**, will consist of a question-and-answer period, in which all of the previous speakers will be available for further comment.

Seminar VI, **Ear Training**, will comprise a concert of un-amplified music by faculty and student artists of the University of Miami School of Music.

Seminar VII, **Standardization: Is it Time?**, will be presided over by Ken Pohlmann, with speakers: Almon Clegg of Matsushita, and Bob Younquist of 3M. Ken Pohlmann will present a brief description of the recently announded DASH (Digital Audio Stationary Head) format, which has been adopted by several manufacturers. The adoption of a unified format by several manufacturers may bring the industry a little closer to a de facto standard, which may eventually be followed by more formal standards.

Seminar VIII: **CD Or Not CD, Was That The Question?**, will be addressed by Len Feldman and Richard Elen.

Further details of this digital seminar program, for which the registration fees range from $250 to $350, are available from SPARS, P.O. Box 11333, Beverly Hills, CA 90213. (213) 651-4944.

AUDIOTECHNIQUES APPOINTED FIRST DEALER FOR SONY DIGITAL PRODUCTS

In a recent move that reflects, according to Sony, the growing importance of digital recording technology in professional audio, Audiotechniques has been appointed the first of several new direct sales representatives for the company's range of PCM products. The Stamford, Connecticut-based company, headed by Ham Brosious, will represent the entire line of Sony digital equipment, in addition to its continuing role as dealer for MCI/Sony consoles and tape machines. In the photo below, taken at the recent contract signing, are (L to R): Ham Brosious, with Rick Plushner, national sales manager, Mike Faulkner, regional sales manager, and George Currie, VP and GM of Sony Professional Audio.

MATSUSHITA, MCI, SONY AND STUDER AGREE ON DIGITAL RECORDING FORMAT

All four companies will support a new common format in stationary-head recording called DASH — Digital Audio Stationary Head. The new format combines features of the original format jointly promoted by Sony and Studer, with new developments from all four companies.

The new format agreement takes into account recent developments in technology, including the possible use of thin-film heads, a format for low-speed recording with increased robustness in signal processing, and the recommendation of the AES Standard Committee on Digital Audio for standardization of the 48 kHz sampling frequency.

DASH's specifications and features will be submitted as a proposal for an international specification, and also actively promoted among the manufacturers and users of digital audio as the recommended format for stationary-head digital recording.

The DASH format covers a wide range of application from two-channel (7½ IPS, ¼-inch tape) to 48-channel machines (30 IPS, ½-inch tape), and has three track/channel versions depending on tape speed (fast,

All photography by KATHY COTTER

Session Producer ...　　　**Studio Owner ...**　　　**Label President**

SPENCER PROFFER

Interviewed by David Gordon

Transformerless energy and well-earned pride coursed through Los Angeles' Pasha Music House — *Metal Health* by Quiet Riot had just hit #1 in the charts, to become one of the most successful debut rock albums. As a Platinum "metal" album, it had the added distinction of shaking up the old foundations of the record industry. Quiet Riot has abruptly put Pasha, and its producer, studio owner and label president, **Spencer Proffer**, in a very powerful position. Pasha has a long, dues-paying history, and now it is a hot entity; it means Quiet Riot; it means two strong tracks from the soundtrack of *Staying Alive*; it means a tune in the film, *All The Right Moves*; and the long-awaited Vanilla Fudge reunion album.

How did Proffer slip into such an enviable position? He began as a songwriter for A&M at the age of 17. Gary Lewis and the Playboys recorded his song "Picture Postcard," and by the time he was 20 he had over 70 recorded songs. He recorded as an artist for ABC/Dunhill, MGM Records, and CBS Records. A successful relationship with Clive Davis led to a position on the CBS staff. Meanwhile, he was pursuing studies at UCLA, and went on to law school. In 1974, he became national executive director of United Artists, where he produced 11 Top 50 hits in 18 months, and got industry attention for his progressive production of Tina Turner's *Acid Queen*.

After leaving UA, Proffer formed his own music company and received critical acclaim with Allan Clarke, lead singer of The Hollies. His production of serialized concept albums with Australian success Billy Thorpe, and a touring laser show, exemplified the pre-occupation at Pasha with big productions, bold visuals, and a departure from the usual visual support for recording artists. It's no surprise that Spencer Proffer supervised the Quiet Riot videos, and is energetically courting profitable marriages with the film industry.

R-e/p (David Gordon): How did you first come in contact with Quiet Riot? What made you single them out as a potential Pasha act?
Spencer Proffer: In January 1982 I met Kevin DuBrow who, along with Randy Rhoads and Rudy Sarzo, were the founding members of the band. I met Kevin while picking up a friend of mine, a manager named Pat Armstrong, who handles Molly Hatchet, amongst other

bands, and they were discussing potential management. Kevin and I just started talking about music; he sings in the tradition of some of the classic English rock singers. And that's my favorite genre of rock and roll music — the days of Humble Pie, Deep Purple, Sabbath ... early Rod Stewart. So I immediately asked Pat if I could take a listen to the tape that Kevin left with them. Upon hearing it, I thought, boy, what a classic

voice, reminiscent to a degree of Noddy Holder when he was in a band called Slade. Having heard the tape, I was real keen to see what Kevin and his band were all about live.

At that time, the band lineup included Kevin and drummer Frankie Banali; Carlos [Cavazo] the present guitar player wasn't yet in the band, and Rudy Sarzo was still playing bass with Ozzy Osborne. After seeing them perform —

SPENCER PROFFER

watching Frankie play drums, and Kevin sing — I really felt that this is something that would be tremendous in front of a large audience. I saw a lot of distinctiveness in Kevin's visual and movement approach to rock and roll, as well as some of the best drumming technique from Frankie. So that was really step #1 in our collaborative process.

R-e/p (David Gordon): At that point did you go into the studio and start working on demos for the band?
Spencer Proffer: That was the catalyst for me to take it to the next step, in the true sense of artist development. I didn't want to commit them to me, or me to them, without seeing how the work experience would translate in the studio context. Being fortunate enough to have my own studio, I was able to take some time over a weekend and go in with the two of them. At the time they had another guitarist and bass player, but it was really the foundation of Kevin and Frankie, and the cooperative spirit in which we worked, which made me say, "Yeah, I'd like to pursue this relationship."

R-e/p (David Gordon): What contribution do you think you made to the band, in terms of maybe molding their music?
Spencer Proffer: Well, subsequently Carlos joined from a local metal band called Snow, when we decided that we needed a different guitar player; he also came to the band as a collaborative writer. So, in listening to all of their material — both Kevin's, and Kevin's and Carlos' — they truly had a sense of what the kids would react to. I sensed that one thing that was missing in a lot of the Heavy Metal being played on the radio was the sense of melody, the sense of *true* song construction. And a real sense of audience participation; metal bands are primarily live, exciting, visual bands. The thing I felt most excited about with Quiet Riot was the musicianship of everybody involved; their sense of truly knowing who their audience was; and having music with a melodic flow.

I really just helped nurture and encourage that. I worked with them on some of the arrangements, and heightened that sense of participatory rock, which is what I would call the Quiet Riot "musical genre." It's melodic participatory rock, not necessarily Heavy Metal, although with Carlos' guitar playing, Rudy's bass playing, and Frankie's drumming, it certainly competes as hard and as heavy as anything out there.

R-e/p: Do you feel that you were adding something to the typical Heavy Metal form with Quiet Riot?
SP: Oh, absolutely. I thought that we were really expanding the genre. We were adding an element of melody, but with enough lyrical content and story line so that the kids would enjoy the record once they heard it, and be motivated to go and see the band. We try to have the vocals and the ambience of the record sound as live as possible, so that when you heard it on the radio, first you want to get the feeling of: "Who was it?" and "Where are they playing?" and "When can we see them?" Fortunately,

the record has been the #1 most requested album in this country for about the last 12 consecutive weeks [mid November], so that we happened to strike it right on the money with that approach.

R-e/p: How did you achieve that live, open sound in the studio?
SP: By not going for a lot of overdubs. For the most part, we recorded the album, using our entire building [Pasha Music House] as a studio — not just our control room and actual studio. We ripped apart our lounge, which is all brick and wood, and put Carlos' amps in there. We ripped apart our second studio, and put Rudy's amps in that. We had the ambience of the drums as the only sound in our main studio. By using separate environments we gave the album as much of an open-air, large feel as possible. There was a lot of reflective wood and brick on the walls.

It didn't take us more than five weeks to make the Quiet Riot album. We didn't approach it in a very sterile sense, but rather in a very aggressive, live-feeling sense, so that sonically it would give you the feeling of being out in a large stadium. While working on the vocals, I didn't restrain Kevin; I tried to push him even harder, to sound as loose and as natural as he is on stage.

R-e/p: I assume that there'll be a second album.
SP: There will be a second album that we'll be recording in January, which will be entitled *Condition Critical.*

R-e/p: Do you plan to make a departure from the sound of the first album?
SP: Not at all. It's going to reinforce it. We'll retain everything we had, and might be a bit more adventurous but, by and large, it's going to retain the same "melodic anthem," participatory feel.

R-e/p: Let's move on to you personally. It's not too common these days for one individual to have so many hats in his closet; you're a studio owner, producer, and you have your own label. Do you also get involved in the engineering side?
SP: Oh, I get involved in it. I've got a first-rate engineer in Duane Baron, who's been with me now for six years —he actually started his career at Pasha. But I'll mix my own records.

Duane's been working with me long enough to know what sounds to go for, relative to the nature of the project we're working on. But, insofar as putting the records together, I need to have that *total* involvement in order for me to be called the producer. If I turn it over to someone, I can't feel that I am justifiably doing my job. So I do get involved on a very broad context in the music and the arrangements, and certainly in the way the records ultimately are put together, which is the final mix.

When you mention my wearing various hats, all of those are support mechanisms to my first and foremost love and function, which is to make records. I built my own studio because I wanted to have the tools necessary to make state-of-the-art records.

R-e/p: When did you build Pasha Music House?
SP: In 1978 . . . and literally mortgaged everything that I had in order to do it! But I don't like looking at a clock when I work, and I don't like going places and having certain things not functioning on the level that I would choose to deal with. I felt that if I had my own facility, which was built to complement my style of recording, and the way that I make records, then I would have an edge in at least offering every option, sonically,

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66 **Having grown up as a songwriter and performer, I think that sometimes the best producers and writers are those people that really have an understanding of the *entire* medium.** 99

DRUM MIKING:
KICK: MD421 (INSIDE)
U87 (OUTSIDE)
SNARE: SM57
TOMS: SM57
O/Hs: KM88

STUDIO B

BASS AMPS

U67

CONTROL ROOM B

SCRATCH VOCAL

RE20

• U87 (DISTANT)

• 435 (CEILING)

U87 (DISTANT)

U87

U87

MD421

MD421

DI

PZM

PZM

OVERHEAD

THREE-LAYER GLASS WINDOW

LOUNGE

NEUMANN U67 ON HIGH BOOM

CONTROL ROOM A

GUITAR AMPS

TUBE RIBBON MIKE

U67

PASHA STUDIO LAYOUT FOR QUIET RIOT BASICS

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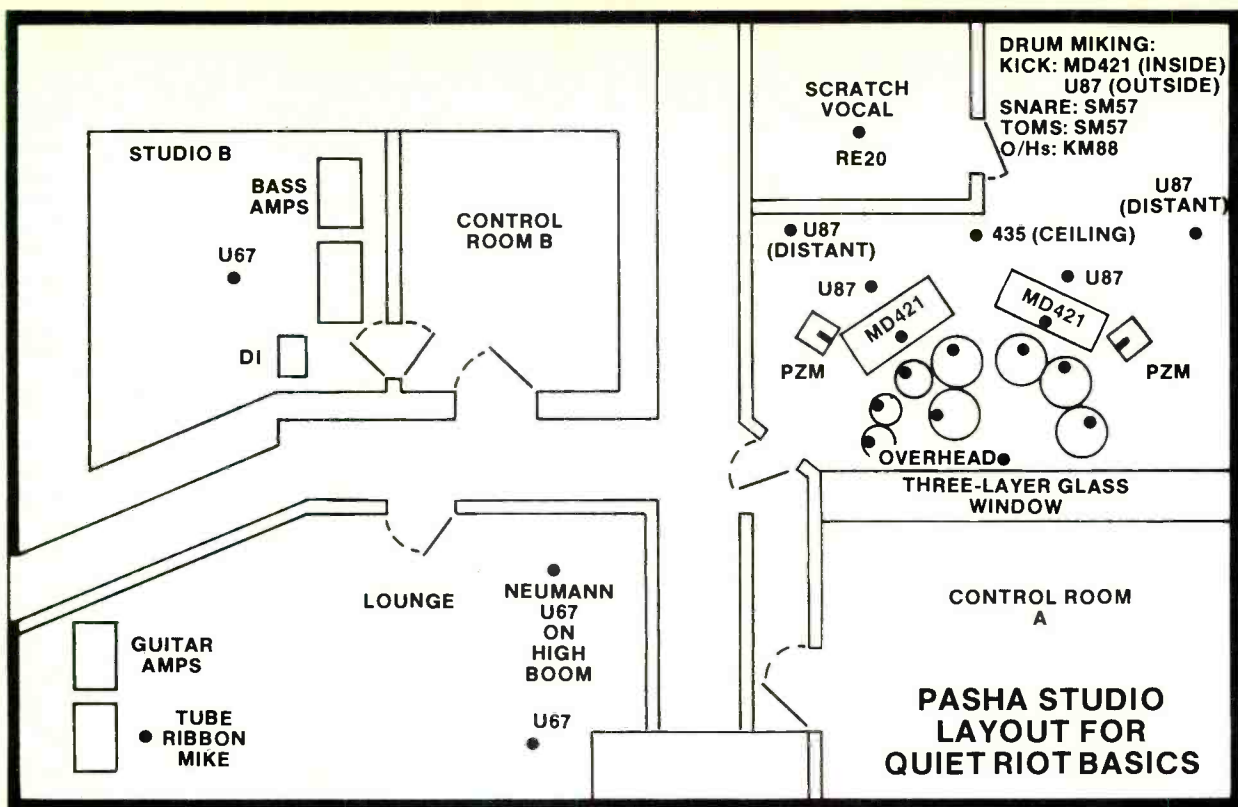
aesthetically, emotionally, and practically.

R-e/p: Is there any special sound or facility aspect you were after in building your own studio?
SP: The monitoring is real true. I'll give you an example. We master our records with George Marino, who's with Sterling Sound in New York. George happens to be, for me, *the* finest mastering engineer, but he has an easy job with our records. Everything we put on tape requires very little, if any, equalization going into the translation to disk. The room is *accurate*.

When Larry Brown, who is a fine engineer, and I built the room in 1978, we took great pains to make sure the monitoring was true. The room felt like the kind of place you could spend several months making a record.

Aesthetically, the room is called The Pasha Music House, and our whole building is meant to feel like a *home*. People spend a good deal of their lives here, so the ambience of it is like being in an English country house, which does set it apart from the normal "commercial feel" of most studios I've been in.

R-e/p: Before we get on to the intraworkings of the Pasha family of companies, why did you decide to form your own record label?
SP: To have a little more control over the ultimate destiny of my work. I found that having had some experience working for record companies — both at CBS, and running A&R at United Artists — I could see some of the system's shortcomings. Having had the benefit of that experience, no one understands the music better than the people who make

it. Fortunately, having some music-industry background, I have ideas about the marketing of the music I'm involved with. The logical corollary of this experience would be that if I found something I believed in, to then make use of a major company such as CBS, with its marketing, promotion, and distribution resources, thereby allowing me to make some of the creative judgments.

With the way our label is set up, CBS is my partner, but they do give me the latitude to find an artist; handle artist development; bring the project to them in a recorded form; and give them some ideas both in the marketing, merchandising and promotion. When you're *just*

PASHA STUDIO A CONTROL ROOM EQUIPMENT LIST
Console: Transformerless MCI JH-500 Series with JH-50 Automation Package.
Monitors: Big Reds with Mastering Lab Crossover, Yamaha NS-10Ms, and Auratones Sound Cubes, powered by Yamaha P-2200 and P-2050 amps.
Multitrack: MCI JH-114 24-track.
Tape Machines: MCI JH-110A two-track, Studer A80 half- and ¼-inch two-tracks, and Sony TC850 two-track.
Reverbs/Delay Lines: EMT 140 plate reverb, AKG BX10 and BX20 spring reverb, Fairchild 658, Cooper Time Cube, EMT 251, and AMS RMX-16 digital reverbs, Lexicon PCM41 and Prime Time, Eventide 1745M and H-949 Harmonizer, Marshall Time Modulator, B&B Audio EQF-2, Orban Dynamic Sibilance Controller and Stereo Synthesizer, plus Aphex II Studio Aural Exciter.
Compressor-limiters: Teletronix LA-2A, UREI LA-3A, 1176 and 1178, Valley People Gain Brain and Kepex II, plus Inovonics 201.

a producer, you might have a lot of good, valid ideas, and the label can say, "Go make your records, and keep your mouth shut." At least with a label identity, and the staff to back it up, we can present ideas, campaigns, concepts, and video ideas that, if the major agrees — and fortunately CBS has been wonderfully cooperative with us — we can then keep a continuity of image and profile that the artists want to have when they make their music.

So I just get involved in the conceptualization, but I move on and continue to make records while my organization does the follow-through consistent with the ideas that the artists and I have about the music. It's relative security —there's *no* security in this business anyway — but at least I know that when I finish a record it's going to be handled in a manner consistent with the spirit in which it was made.

R-e/p: When you're developing an act, do you take those demos or pre-production forms to CBS for approval?
SP: No. They have pretty much given me the latitude. There are some very musical people at CBS that I have a lot of respect for but, from an A&R standpoint, it's pretty much my call. Tony Martell, who is most responsible for bringing Pasha to CBS, is the vice president of CBS' Associated Labels, and has been very much the guiding light for me at CBS, together with Don Dempsey and Walter Yetnikoff, who blessed the whole deal at the top corporate level. Tony has been tremendously supportive on every level, and I find it a pleasure to submit my ideas and my projects to him as a sounding board.

SPENCER PROFFER

R-e/p: Someone once referred to Pasha as a true "boutique" record company. Can you elaborate on that description?
SP: Boutique is a clothing term. When you go into some clothing stores, you've got to buy off the rack. When you come to Pasha, you have it *custom tailored.* And some of our projects are not necessarily on the Pasha label. We make some records for other record companies, if I'm very motivated about the project and the artist.

R-e/p: What are the main differences when you're handling an outside project like that?
SP: We make less money! We also have somewhat less control, although most of the major labels we're dealing with welcome out contribution. Fortunately we get involved in some of the follow-through areas, but we don't have as much to say because they aren't our artists that we've developed from the ground up. But I offer that as an adjunct to my creative services, so that the time I've invested is somewhat protected and accentuated by our ability to get some records on the radio, and set up the campaigns in conjunction with the management and the respective labels.

R-e/p: What was your first job in the music industry?
SP: I was a songwriter, and I wish I could have gotten a job being one! I was working my way through UCLA in a band and, through a series of events, wound up having a number of my songs recorded, the first of which was "Picture Postcard" by Gary Lewis & the Playboys back in 1967/68. By the time I was 20, I had about 100 of my songs recorded by various artists throughout the industry.

R-e/p: What were you studying at UCLA?
SP: I was pre-law. I had two majors actually — political science and music — and was learning how to orchestrate for symphony, and trying to hone down my mind to go to law school, which I ultimately did.

R-e/p: Has the fact that you are a qualified lawyer helped your career in the music business?
SP: Well, it certainly allowed me to receive what I was *contracted* to receive! It's made it easier for me to become involved with projects, and conclude them without a lot of red tape. I've been able to understand the business parameters, and how they actually work in an operational sense; I have to *live* my deals. Most lawyers make and negotiate the deals, and then walk away from them. By being able to up-front negotiate the deals that I and my company have to live with, they become more realistic, more functional, and

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there are less potential legalities to present problems. I've never had a legal problem in any contract that I've ever signed in my career! I've found that to be a tremendous aid in getting more business concluded quicker, so I could spend more time on the music and not get hung up on the red tape.

R-e/p: Let's talk about a few of what you personally consider to be the major breakthroughs in your career over the previous decade or so. One that comes to mind is the album Acid Queen.
SP: That brings to mind a really fertile period in music, back in 1974/75, when I got involved producing Ike and Tina Turner. Tina was cast as the Acid Queen in the Ken Russell movie of Pete Townshend's rock opera *Tommy.* I felt that this particular role would be a great broadening device for Tina's audience; she would be seen by people who loved The Who, and loved contemporary con-

cert rock. Since Janis Joplin there hadn't been a female rocker who really kicked ass, and who could take a rock song and interpret it. I thought Tina to be one of the finest interpretative singers of our generation. I had just produced a single for Ike and Tina prior to this movie coming up, and thought that it might be pretty exciting to take a voice like Tina's, couple it with songs by people like Pete Townshend, Jagger and Richards, and open up her audience to a much wider demographic.
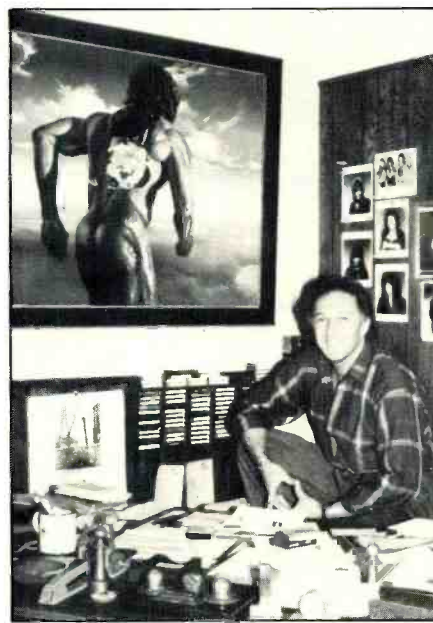
I found that album be be a very exciting project, because it fused her style of singing with progressive, melodic rock, and was one of the first projects I had ever coproduced that went a little left of center. *Acid Queen* wasn't quite your normal commercial record, although its novelty itself became commercial.

Sometimes people lead, and sometimes people follow; my own personality has a tendency to want to take a step forward, and not necessarily look over my shoulder to see if I'm copying any-

body. I'd rather try and *set* trends, than follow them. That was really my first opportunity to take one form of music, integrate it with another style, and achieve something that would create a new appeal.

R-e/p: Is there anything about the sound of Acid Queen *that you consider to be a step forward in terms of production?*
SP: Well, it was more the coalition of her style and rock and roll. I had some great musicians involved on that album: The Crusaders' Wilton Felder and Joe Sample played keyboard; Ray Parker, Jr. and myself did the guitar work. The sound of it was technically as good as records may have sounded in that day. My standard of sonics was always one of creating some drama and adventure on record, so I tried to make it sound exciting and the kind of record you could actually visualize.

R-e/p: Let's move on to the Allan Clarke solo album project, including Legendary Heroes. *Was that another involvement that altered the course of your career?*
SP: It was an interesting collaboration. I grew up listening to the Hollies, Beatles, Stones, and The Who. In terms of the manner in which they structured their harmony, Allan Clarke, Graham Nash, and Tony Hicks of The Hollies were a very influential component of my own musical development. Having a chance to do some of his solo records was very exciting.

I remember he and the Hollies were the first people to record Bruce Springsteen material outside of Bruce's own recordings. Allan and Bruce had developed a relationship, and one of the first songs that I recorded with them, cut #1, side one of Allan's *I've Got Time* album was a song called "Blinded By the Light."

When I delivered that album, which had original songs by Springsteen, Carole Bayer Sager, Melissa Manchester, Nicky Chin and Mike Chapman, the record company [Elektra] did not agree that "Blinded By the Light" was the kind of material that would be played on the radio. They thought the lyrics were too obscure; that Allan would be better off doing other kinds of material. As a result, another English band, Manfred Mann, did a version of the song not far away from our version, and it went on to be the biggest record of 1976.

A little heartbreaking from our end maybe, but at least it vindicated my own belief that [it would work if] I took a voice like Allan's, put it into a different genre with writing like Bruce Springsteen's, and did something that I felt was true to what Allan deserved to record. We did subsequently have a hit with a song called "Shadow in the Street," which made the Top 30. I found working with Allan just a wonderful, enriching experience.

The next male artist that I had gotten
... *continued on page 51* —

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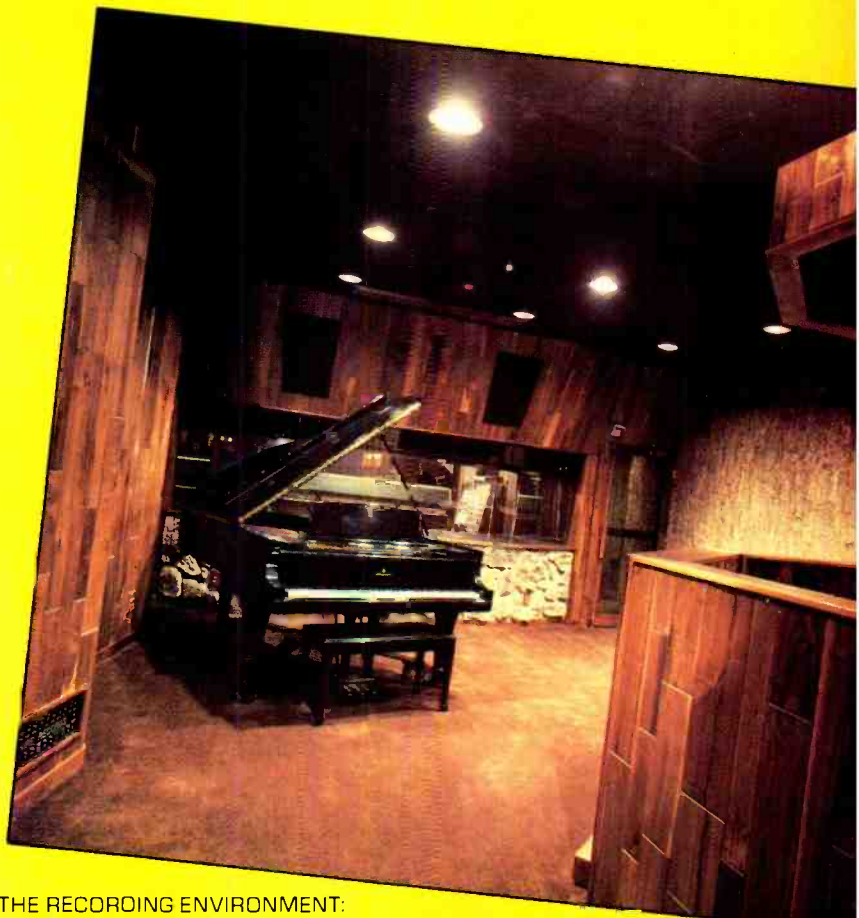
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involved with was an Australian "superstar" by the name of Billy Thorpe. Allan Clarke and Billy had been friends when he lived in London. Billy actually came in and played a guest solo on one of my later Allan Clarke albums, which was the beginning of my collaborative, cross-pollination idea of having Pasha artists work together with one another.

R-e/p: How would you describe the way in which you helped shape Billy Thorpe's career? As maybe an example of your contribution to the overall concept of his records, performance, touring, tie-ins from record to record; a long-range type of thing, not just a quick shot.

SP: Well, I can't take all the credit for that. Billy is one of the brightest human beings, both intellectually and creatively, that I've ever met. He came here from Australia, where he'd had 15 #1 records and 10 Platinum albums. When we decided to work together, we really wanted to come upon an approach to break his career in the U.S., so that he wouldn't have to start over totally. We really spent a lot of time getting to know each other, and establishing a pulse on what was happening in society, and how we could make an impact on that

> **❝ I like to make memorable records that . . . are very positive.❞**

with music.

The seminal event that charted our course together was going to see *Close Encounters of the Third Kind*. At the same time the entire space phenomenon had hit the world in the broadest sociological sense. *Omni* magazine became a well-known publication; *Star Wars* and *Battlestar Galactica* went on television. There was a space orientation that pervaded almost every form of life, but yet

no one in the industry hit the "celestial sphere" in music. Okay, they hit it with album covers — ELO, for example, put a spaceship on their cover — had laser shows, but they were singing songs that dealt with *other* subject matter. "Last Train to London" was an ELO song on a space-oriented cover. Boston had a tremendously successful debut album, but they sang love songs — yet the band put the space theme on the cover. So we thought it would be really hip to take the last scene in *Close Encounters* one step further, and tell a story on record: the birth of the Children of the Sun.

R-e/p: What was that story line?

SP: In the year 1991, the Nostradamus prediction of the Eastern and Western powers joining to fight a middle power, and causing the world to blow up, created a shift in the earth's axis. Half the world was disintegrated, and the other half was burning up as the earth's orbit shifted closer to the sun. The Children of the Sun were a friendly race from another galaxy who'd been watching man's self-destruction since the beginning of time on earth, and they came to earth in a very friendly fashion and offered everyone a choice of staying or leaving. And, of course, by the end of the first song on the *Children of the Sun* album everybody split; they figured there was new hope.

The overall theme of the project, and of Billy's recordings, was very hopeful, very positive.

It's truly a fantasy, a movie in sound. Billy and I spent nine months making the first record, in the truest collaborative sense. We co-wrote the song "Children of the Sun," and from there it spawned Billy's fertile mind into creating this whole story.

R-e/p: How did you support such complex space theme music, in terms of sound to augment the basic concept?

SP: A lot of drama. *Children of the Sun* opened up with three minutes of sound effects and ships flying from left to right. We recorded them 46-track by linking two 24-track machines together, so we would have optimum separation and sonic brilliance in our stacked vocal harmonies. But we created sonics; in *21st Century Man*, the follow-up album to *Children of the Sun*, we had an atomic bomb burst that lasted two minutes. To create the effect, we used 24 banks of synthesizers, and it rumbled speakers. Many stereo shops throughout the West Coast used that album as a demo for their sound systems, because if it could withstand all the transients that we had on the record, you *knew* you were buying a good system!

R-e/p: And you played around quite a bit with stereo panning and effects, didn't you?

SP: Oh, yeah, very much so. We had things coming in and out, left to right. We had something coming in for four bars that you never heard again 'til the

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QUIET RIOT AND VANILLA FUDGE SESSIONS
— A Conversation with Pasha Engineer Duane Baron

R-e/p caught up with engineer Duane Baron during a typically hectic day of sessions at The Pasha Music House. An afternoon of basic tracking with a Los Angeles band, Pictures, called for drums, keyboard, bass and guitars to be set up in the main Studio A recording area. Mike lines and cue feeds were run down the adjacent hallway to the lounge, where dual Marshall guitar amps had been set up in the brick-lined room. To intensify the degree of reflective ambience, Baron had placed sheets of plywood underneath and around the guitar amps, and

Spencer Proffer and Duane Baron

had miked them with a vintage ribbon mike close to one of the cabinets. A pair of Neumann U87s mounted on high booms also were placed 10 feet from the amps to pick up room ambience. With the proven chart success of the Quiet Riot album, *Metal Health*, and the anticipated success of the Vanilla Fudge reunion album, titled *Mystery*, and due for release early next year, we asked Duane Baron to elaborate on his particular recording and production techniques.

"We wanted an 'expansive' sound with Quiet Riot and The Fudge," he explains, "and wanted to keep Jeff [Beck] happy. He oversaw everything that we did. I didn't really do anything different with him than I do with most guitarists I work with."

Jeff Beck's lead guitar tracks posed a considerable challenge because scheduling required that they be recorded mono in England. Session producer Spencer Proffer felt that the original tracks weren't in keeping with the overall sound of the album, and worked with Baron to re-record them. To recreate a more live sound, the mono guitar track was fed to a Marshall amp placed in the studio, and a ribbon microphone placed close in front of it. In addition, two room mikes were used to capture stereo ambience. The feed from the close ribbon mike was split to two input channels on the board, one of which was routed through an Eventide H949 Harmonizer. "I used the Harmonizer to split the pitch a bit on the right," Baron recalls, "and then I added the room mikes in stereo. You don't really hear the Harmonizer; the effect is blended with the room mikes, so you get the illusion of hearing the true size and sound of a *very* big guitar."

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next album. It took me back to the days when I studied classical music; you'd have motifs for different characters. And there were sound effects that you would hear only briefly, and then would reappear, or certain harmony structures. On *Children of the Sun*, the song "We Welcome You" had the children of the sun welcome the earthlings onboard the ships. We used the exact same vocal harmony structure and approach on the next album during a similar encounter. Those are subliminal links that only the truest fans of the project might pick up on, but we were real proud of it.

R-e/p: Is there a third album in the series?
SP: Yes. It's the unfinished part of the trilogy. Pasha is going to re-release *Children of the Sun* in 1984, the "George Orwell" year. The album sold about 400,000 domestically after nine months of shopping it. Everyone told me I was crazy, that the kids would never relate to it, yet it became one of the most requested albums in 1979. To coincide

with its re-release, I'd like to do a video, which we were not permitted to make at the time of its original release because visual-music was not "in."

R-e/p: Before moving on to your involvement with video, let's talk about the stage show you mounted for Billy Thorpe. That was pretty progressive at the time, wasn't it?
SP: Well, not only was the particular performance aspect of Billy's show very visual, but we also had a touring laser show in 1980 that choreographed the storyline for both *Children of the Sun* and *21st Century Man*. We ran that show in planetariums all over the world to really encapsulate audio-visual entertainment. It was too expensive to mount the laser show on video itself, although we approached RCA Videodisk and a number of other companies, and said it would represent sufficient abstract entertainment to warrant tremendous repeatability, *if* we could get it on the air. But it proved to be too expensive a process to put on the air, so we just toured it in a way that would have been cost-prohibitive to do with the band. To have the touring planetarium show run in nine cities simultaneously, however,
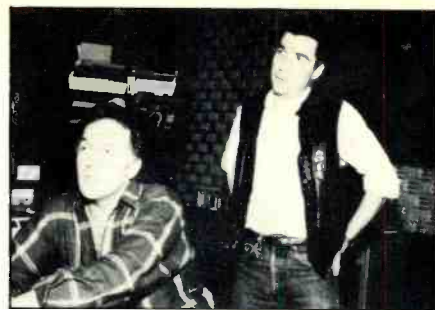
Sessions at Pasha Music House with Rod Falconer

together with cross-pollenating marketing of the records, we found to be a very exciting merger of the music and visual mediums.

We were a bit ahead of ourselves in that a number of people thought we were crazy doing this too. They said, "Marrying audio and visual? No way. Have hit singles!" There were people that really thought we had jumped off the deep end. I mean, I was telling them I wanted to put this on Broadway, in the true sense of combining the mediums. And they just kept screaming: "Hit single! Hit single!" And this is the kind of project that had more *depth* than just necessitating a hit single.

R-e/p: Let's take this visual pre-occupation into the world of music video, which is something you're involved with now. Quiet Riot's latest video is creating a lot of response.
SP: Well, in the Quiet Riot videos we managed to be one of the first people to even link a couple of videos together. In our first video, on the title track, "Metal Health," we have our hero, with the mask, which also is on the album cover. I'm a strong believer in visual identification with the package, and with the album, so we had the cover come to life at the beginning of that video. As Kevin [DuBrow] breaks out of the padded room into the mental institution, we had a reason for his performance — he needed to break out so that he could get on-stage with the rest of the band. And when he saw a flashing light in the corridor of the hospital, he jumped down a large hole that landed him on-stage, and he threw the mask into the audience.

In our second video, the kid who caught the mask from the first video is lying on the bed, with the mask hanging over his bed in very much of a religious fashion. We actually put a flashback in our second video to tie it together with the first.

The director of that video, Mark Rezyka, and Steve Einczig from CBS, Warren Entner, the band's manager, and myself really brainstormed how we could tie all these elements together, so that there would be a continuity to the whole Quiet Riot visual as well as sonic stamp. We plan to use the same illustrator that did the cover on our second album — to incorporate elements of the mask from the first one — because this

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QUIET RIOT AND VANILLA FUDGE — continued . . .

What made him select a vintage tube ribbon microphone to record loud amplified guitars, we queried?

"I've tried all kinds of mikes," Baron says. "I get old ribbon mikes from radio stations. People think they are junk, but I've found that they work best for me. Other engineers feel that I'm crazy, because you can blow them out if you put ribbon mikes in front of Marshalls. And yes, after a while, they are only good for guitar; the ribbon stretches out. But, when the pressure of those Marshalls hits them, you get a *killer* guitar sound. I have to admit I've felt guilty putting these old suckers in situations like this."

Any particular choice of vocal mike? "We set up an assortment of microphones: usually a Neumann U47, a U49, a U87, and AKG C12, C414, and C24. The singer will try all of them, and Spencer and I will listen and decide which one works best for the particular voice. We test old mikes before we buy them because it's like buying a used car; just because it's a Ford doesn't mean that it's going to be a *great* Ford. You take them for a test drive, and a lot of them don't sound good. But then you find one that still has its old quality. The AKG C12 is one of my favorite vocal mikes. We used it a lot on the Vanilla Fudge sessions."

Baron began his career as an assistant engineer for Larry Brown, who also had a hand in the design of Pasha. Studio A is comprised of a large recording room with drum booth, and a spacious control room. Studio B is smaller, and used mainly for overdubs and vocals. With the recording of groups such as Quiet Riot, additional areas of the studio complex have been utilized as recording areas.

"For basic tracks," Baron recalls, "we have instruments everywhere. The rooms were originally designed for MOR work, and were relatively dead sounding. We have added a lot of reflective surfaces, and expanded into rooms like the lounge for versatility. The *whole* building has become the studio."

An unrestricted approach to recording, and a solid relationship with Spencer Proffer, has resulted in a production team that suddenly has leapt into the limelight. What was Proffer like to work with in the studio?

"He lets me do my job," Baron says. "The great thing about working with Spencer is that he gives me the ball; he lets me go. He will give me an idea, describe what he is looking for, and then leave it up to me. The more we work together, the more he can say just one word and I'll know what he wants.

"There is a big advantage to working with a producer who owns his own studio — I have the time to take an idea as far as I can go. He will leave for a few hours, and after I've burned my ears out with the sound, he'll come back with fresh ears, listen, and make further suggestions. He trusts me enough to let me work on a sound on my own, and then come back with fresh ideas. It's a good working relationship, like between a quarterback and a wide receiver."

With just an hour between sessions, Baron had to make some rough mixes of the basic tracks for Pictures, break down their instrument setup, and lay out the studio for an evening session with Rod Falconer. "Doing two drum setups in one day takes quite an effort," he comments breathlessly, as he heads back to work. ☐☐☐
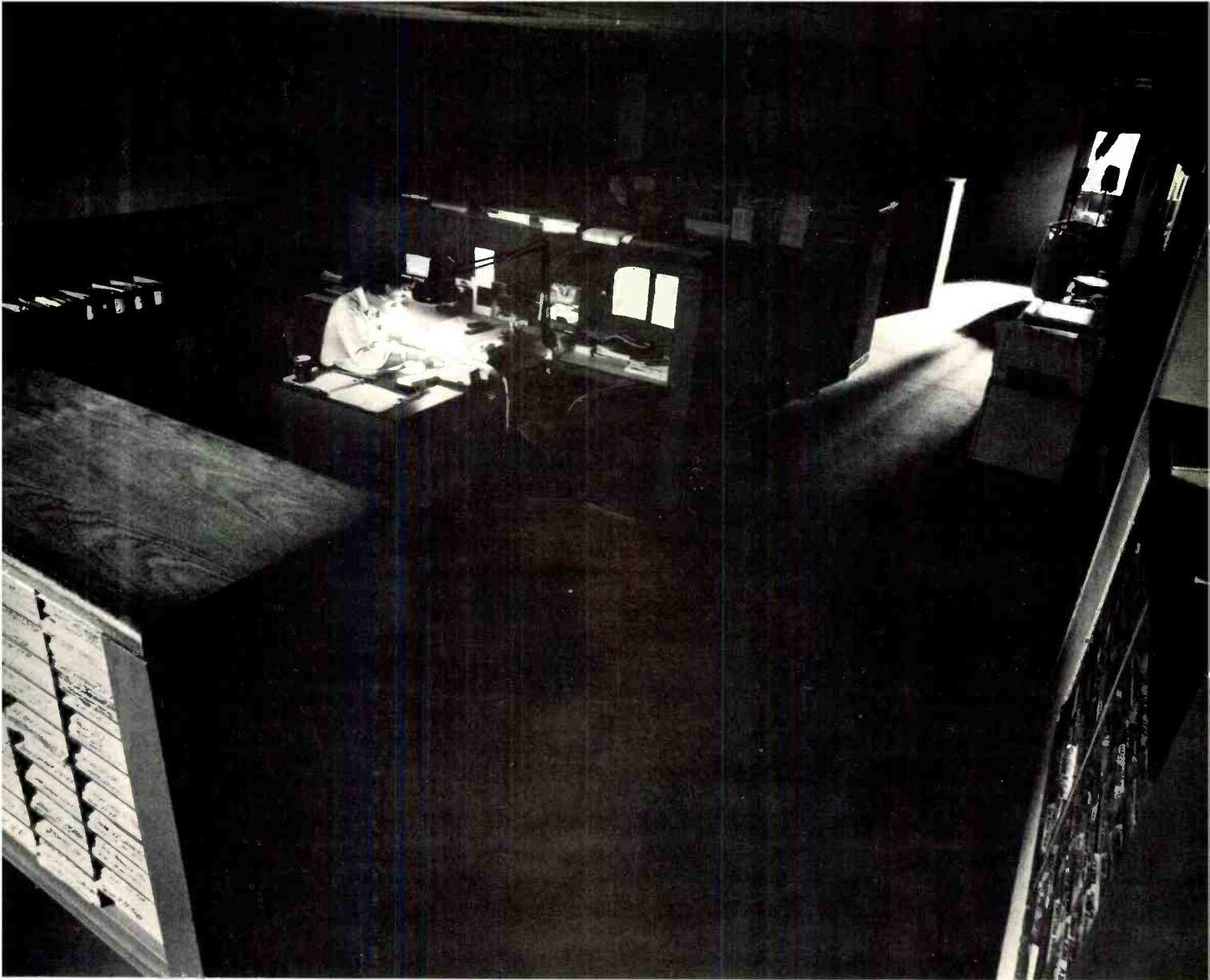
Like all of our people, Mike is a self-starter.
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gives the consumer a continuing link into staying with an artist. The records won't be a one off; the videos won't be a one off.

R-e/p: While producing an album project do you have a visual in mind as you're laying down the music tracks?
SP: Very much so. Most of the records that I've made tend to have a lot of drama to them. And I also look to get involved with projects that have a very *visual* sense. I'm making a record now which is a true pioneer of the collaborative mediums. Roderick Falconer is currently directing and writing a screenplay. He wrote the screenplay to *Star Chamber*, which was released last year, and now is going to be writing and directing a film called *Empire Man*. So that for a visually orientated guy like Rod, all his songs are "mini-movies." Every time we finish working on a piece for the record, we talk about how that's going to translate into video. I would very much like to do a video of Rod's entire LP, because a lot of the songs have a storyline that we've weaved together.

R-e/p: We're really talking about the many different, supposedly separate, forms of art — music, records, live performance, music-videos, and, of course, film. The connection is very obvious with artists you're now working with. How will Pasha become involved with the film industry?
SP: One way is certainly the proliferation of our involvement in videos that portray the music that comes out of here. The next step is that in 1984 we are going to be doing at least three complete movie soundtracks.

R-e/p: You had an involvement previously with a couple of tracks for Staying Alive.
SP: Yes, we have two tracks on the *Staying Alive* soundtrack, outside of the Bee Gees and the Frank Stallone tracks. That came about through a series of relationships, and my desire to get involved in the film business. We have another track in *All the Right Moves*, the new Tom Cruise film, on which Danny Spanos is singing. That was a song tailored for the film, and recorded specifically for emotional impact in the movie.

Something I very much look forward to doing is being given an entire script or film, and being asked to work with

the producer and director in picking the moments at which contemporary music would fit in. Then, have the songs written and recorded to be custom-tailored to the movie, but also have a broad enough appeal to stand up as conventional records.

We are now in the process of working on the music for a film for a major studio that we are going to be doing top to bottom, and next year we'll have two other pictures that we have been contracted to do from the ground up. That will still allow me to work first and foremost as a record producer, but then to visually translate the music into correlating with somebody else's visual images.

R-e/p: It might be interesting to take one of the Pasha artists as an example of

this cross-pollination process. Randy Bishop seems to have slipped into various projects. How does a person like that operate at Pasha?
SP: Well, Randy is one of the most brilliant all-around talents I've ever encountered. He is a poet in his own right, but also has a tremendous musical sense. My initial involvement with Randy was producing him as an artist. But he is such a prolific writer, and such a consummate musical talent, that on every project I get involved with there is space for a self-contained element like Randy; for example, the new Vanilla Fudge record, which I just finished producing, and which features Jeff Beck on guitar. Mark Stein and Carmine Appice are two of the finest songwriters that I've come across, but they were not as proficient lyricists as the images of their music dictated. So we needed to find a lyricist to take some of Mark and Carmine's melodies and musical thoughts, and to be true to those very specific feelings that the music evoked.

So, I immediately contacted Randy and had him get together with them.

Initially, there was one tune Randy and I collaborated on that Carmine and Mark wrote the music to. Randy seemed to have such a fine sense of where the music was going that when Mark and Carmine came up with melodies that I felt were right for the album, we turned the melodies over to Randy. He wove beautiful pictures and stories around those melodies that were totally congruent and consistent with the record I wanted to make.

Randy co-wrote the first Danny Spanos single off our *Passion in the Dark* EP: as a matter of fact, Randy co-wrote three of the key tracks on that album, both musically and lyrically. I just find Randy to be tremendously prolific, and a real team player. He is now going to be producing a project for Pasha, for which he will be collaborating on the writing with the band, Pictures, on some of the material. I'll be overseeing it as an executive producer.

I see Randy's role here at Pasha in a very expansive sense, as an artist, collaborative writer, producer, and a true part of our musical family. Thus I am not hesitant at all in involving him, because if I'm afforded the opportunity to work on a project, then people have come here because of a certain style and approach. It's very important to me that whoever else gets involved — where I can't do everything on a project — I can turn it over to someone who has as much musical sense.

Having grown up as a songwriter and performer, I think that sometimes the best producers and writers are those people that really have an understanding of the *entire* medium; I think that certainly Randy is a shining example of that. Carmine Appice is another one of our artists who, as a writer, performer, and potential producer of acts that will be involved with Pasha, can function as one of those total musical entities.

R-e/p: Let's touch briefly on the recent Vanilla Fudge reunion album. What considerations did you have in mind while translating the sound of a band from the Sixties, to a record that would work in 1984?
SP: I wanted to keep consistent the elements that made them big, which was a certain orchestral and dramatic elements. Mark Stein is a very dramatic singer and keyboard player. Timmy Bogert is a dramatic bass player, and Carmine is certainly the most dramatic rock drummer I've heard! So I knew the record had to have *drama*; it had to have a lot of emotional impact, and a lot of color.

Thus, in going through all the mate-

> **" I don't like looking at the clock when I work, and I don't like . . . certain things not functioning on the level that I would choose to deal with. "**

was recorded in one genre by Dionne Warwick, and brought into a heavy, contemporary genre by the Fudge approach.

I'm most proud of that album in its original material, because it's *very* picturesque. The album's called *Mystery*; each song paints its own picture, and tells its own story. I consider it to be one of my favorite records that I can still listen to after having done the work, because it is really fulfilling musically.

R-e/p: What was it like working with Jeff Beck, who guested on the album?
SP: A dream. Being able to work with somebody like Jeff, who very graciously agreed to play as a guest guitarist on two of the tracks, was just magic. He is one of the most lyrical and melodic players I've ever had the opportunity to see, much less work with in the studio.

R-e/p: Let's close with an explanation of Pasha's motto: "Music for people with imagination."
SP: I just like closing my eyes while listening to music, and seeing the images dance in front of me. With Quiet Riot, you close your eyes and envision 30,000 screaming kids getting vibed up. With Vanilla Fudge, you can picture maybe elements of a beautiful Ken Russell or Fellini movie. With Billy Thorpe, you know you're definitely in the zone with Lucas.

I really am a big fan of records that

rial that they were writing, I selected the melodies that were the most dramatic, and yet at the same time ones that I felt were the most contemporary. We did two covers of outside songs that we totally re-arranged to remain consistent with some of the old Fudge material. We recorded "Walk On By," but it starts out with a 60-second orchestral intro that you think would lead into a modern "techno-symphony." But then it bursts into a real good rock feel of a song that

are very *visual*, and so I will continually strive to make records that really give a listener a lot more to go on than just "listening" to a record. I like to make memorable records that also feel good, and are very positive; I don't like producing negative lyrics. I like making records that make you feel good, and want to take a little journey with the artist. That way there's much more repeatability to them. I will strive to continue to make records at that qualitative level. ∎∎∎

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SUPERTRAMP ON THE ROAD WITH DELICATE PRODUCTION CO.

by David Scheirman

Once upon a time (no, this is *not* a fairy tale), there was a man who wanted to design and build the best PA equipment available; there was a sound crew who wanted to use that gear; and there was a band who wanted to make it to the top. Dave Martin, of Martin Audio Ltd., London, England, has earned a firmly-established reputation as one of the first developers of an integrated, fully horn-loaded concert sound system, which helped to displace the old column-type loudspeaker systems commonly used in the early days of rock concert touring. The Delicate Production Company, Inc., of Canoga Park, CA, is made up of several past (and present) crew members of the group Supertramp, and services some of the hottest touring accounts out this season, including Men At Work, and the Little River Band. And the group that spawned the production company, Supertramp . . . whose achievements in the recording business are legendary. To observe this unique combination of gear, technicians, and musicians, this writer journeyed to St. Louis, Missouri, to catch one of the last shows on the band's 1983 North American tour.

System History

In 1969, Dave Martin ventured to London from Australia, hoping to see and hear a good concert sound system or two. He was sorely disappointed — concert sound systems at that time in England consisted of stacked small speaker columns with a usable frequency range of perhaps 100 Hz to 5 kHz, mountains of these columns being driven by small

100W slave amplifiers. Martin, realizing that horn-loading was the way to go after observing an Iron Butterfly concert at London's famous Albert Hall in 1970, started working on the first loudspeaker enclosures that would carry his name. Martin Audio Limited was formed in 1971 to supply the industry with the company's first product, the 215MK1 bass horn. The bin met with immediate success, and resulted in groups such as Pink Floyd commissioning Martin to build large touring systems for their exclusive use. The concurrent development of the Midas range of mixing consoles helped bring about, in Europe and the British Isles at least, the "Midas/Martin," system, which became the standard by which other sound-reinforcement systems were judged.

"We first heard this system at a Pink Floyd date," explains Supertramp's house mix engineer Russell Pope, who has mixed live sound for the band since 1970. "We had all of us heard about this new system which Dave Martin had developed, and went to see and hear it . . . the band, the crew, the whole bunch of us. We went home and talked about it . . . the band decided that, somehow, some way, they had to get one like it. No two ways about it; we *had* to have that system.

"At the time, Supertramp was really totally unknown. We were all taking a big chance, having decided to get serious and just go for it. A commitment was there from the beginning; the musicians, the crew . . . we all wanted to do everything first-class. So we bought the

system.

"We have always used [the system] for years, everywhere we played. It has slowly evolved, as new products became available . . . we have just upgraded all of our power amplifier section for this tour, and the system is now four-way instead of three-way. But it is still essentially a Midas/Martin system.

"Using the same system night after night, having a correct system design to start with, and having the group be totally committed to their live sound . . . I guess if people think we sound good live, it's because of a combination of all of those things taken together."

Ian Lloyd-Bisley has mixed the band's on-stage monitors for more than a decade, and offers similar comments: "I wouldn't use any other speakers. We have Martin LE200 wedge monitors, which were first introduced in 1973, I believe. They have always worked so well for me, right from the beginning, that I have never added sidefills to the system. And I have never even used outboard equalization until this year . . . the boxes sound that good."

Ian Lloyd-Bisley

Bisley started with Supertramp as a stage equipment handler, and began mixing monitors for the group in 1973 when Martin Audio first brought out the LE200 wedges.

"Just looking over there, and seeing me still at the desk, after all of these years," he remarks, "makes the group

have much more confidence in their stage sound. I suppose it is unusual for Russell and I to have been with the same band for so long. In this business, sound engineers seem to come and go every other tour. But this has been my *life*; it's what I do."

"It's very simple," stresses Dougie Thomson, bass guitarist for Supertramp. "We got the *best* PA we could find. Our engineers have been with us from the beginning, since none of us had a clue what we were doing. We have all learned this business together. These guys have honed the sound to a fine edge, kept the system abreast of new developments and, if it has all worked, then that is only because that's what we all wanted to happen, years ago."

Musicians' Involvement

Thomson professes more than just a passing interest in the concert sound system the group acquired, as do other members of Supertramp.

"Before this band started up, I had a degree in mechanical engineering from college, so I was quite interested in hardware and that sort of thing. I have *always* been right on top of the guys, sometimes they have hated me for it," Thomson laughs. "But, our *sound system . . . our live sound . . .* that has always mattered a great deal to us.

"When we first started up, we had this idea of what we wanted to sound like. We started to tour and get a name, recognition, and so forth. We were fortunate to have a record company that recognized our desire to sound good, and purchasing our own system was one of our biggest priorities. And, as we became a touring entity, we sort of turned it over to our crew guys. They took the system, which we looked at from an aesthetic point of view, and refined it — [they] took in all of the practical considerations, putting it up every night and so forth, and now they are off and running with it.

"Last tour, we let them keep the PA and lights over here in the States, to see if they could drum up business with it, and keep working, since we weren't on the road. And now, their company is almost as well known as we are!"

Chris "Smoother" Smyth, a veteran Supertramp crew member who is now a shareholder in The Delicate Production Company, offers his viewpoint: "this band always demanded excellence in all areas. In the early days, they spent a lot of money to get things just right; the sound and lighting, and so on. And a lot of money in research and development with Dave Martin.

"You could say that Supertramp helped to pioneer the whole idea of what a concert sound system is today, particularly [with regard to] separate monitor mixes, hanging the main system, that sort of thing. Basically, these guys had their whole hardware thing together. And we, as a crew, did a whole 10-month tour with everything. We collectively built this system up from nothing, came up with the money to make it work, and everybody *cared* about the system as if it were a baby."

House Speaker System

The Supertramp PA, as supplied by The Delicate Production Company, began as a standard Martin modular three-way, horn-loaded system. Bass bins are the Model 215 cabinet, each of which contain two JBL K140 cones. The bass cabinet is commonly called a "split-bin" by the sound crew, since it features two, independently-loaded horn chambers separated or "split" by heavy ¾-inch plywood reinforcing walls

Figure 1: Delicate Productions' four-way flown system for Supertramp Tour, utilizing Martin bass and mid-range bins, plus Emilar/Renkus-Heinz high-mid and Emilar HF horn cabinets. However, the rig is highly adaptable, and can be re-arranged as necessary.

A = MODEL 215 BASS BIN
B = MODEL MH212 MID RANGE
C = RENKUS-HEINZ MF HORN
D = EMILAR HF HORN

BINDING STRAPS

(Figure 1).

The Model 215 multicell exponential bass horn has a rated frequency response of 35 Hz to 1 kHz. Drivers are loaded in a semi-forward facing configuration, and each cabinet is designed to handle 500 watts, depending on the drivers with which it is loaded. Deflector panels and a curved mouth help to assist in the dispersion of the more directional upper bass frequencies.

The low-mids in the system are handled by the Model MH212 midrange horn. In this cabinet, two ATC 12-inch cone drivers are compression-loaded into a 90-degree fiberglass horn, which has a 40-degree vertical dispersion pattern. The cabinet is said to accurately reproduce audio frequencies from 180 Hz up to 2.5 kHz, due to its flare rate. Each box features DC speaker protection via an in-line capacitor, and a selec-

Figure 2: Speaker system on ground, prior to being raised above stage area.

The amplifiers are driven by a four-way Brooke-Siren Systems electronic crossover at 250 Hz, 1.25 kHz, and 5 kHz; basically the same points recommended by Martin Audio for the speaker system.

"I thought about stopping the mids at 1 kHz," Berg states. "But, I decided not to. You have to watch what you put into the bottom end of the horns. With this system though, our problems with losing horn diaphragms occur not so much from excess voltage, but more from impact damage . . . stage hands setting the boxes down too hard and jarring them; that sort of thing. But that happens very, very rarely."

Berg comments that the 15-inch cone speakers fail only when the JBL K-140s finally "fall to pieces physically . . . terminal lead clips may fall off, the magnet loosens, but only after *thousands* of truck miles does that happen. Our loudspeaker failure problems are invariably due to travel, vibration, and handling — not to blowing them up with too hot a signal."

Berg keeps a close watch on the electronic and acoustical transducer components from his mobile repair bench, which goes to every show with him. "It

Figure 6: Additional house mix inputs were handled by an auxiliary 18-channel Trident Trimix submixer for drums and percussion.

used to belong to Roger Hodgson in the band," he offers. "It was his food case. He used to cook his own food on the road, and it was filled with pots and pans, salt and pepper — that sort of stuff. I acquired it a few years ago, and it's now my base of operations for repair and maintenance." The road case is stocked with an oscilloscope, test oscillator, and spare modules, circuit cards and other parts for the system's amplifiers and signal processing devices.

House Mix Equipment

Russell Pope's mixing station was an interesting study in form and function. Centered around a 36-channel Midas console and an 18-channel Trident submixer, the gear was arranged so that those devices requiring the most attention were close at hand, while those which could be preset before the show were out of the way (Figures 5 and 6). The main system drive rack was situated only inches away from the main console, and contained the Brooke-Siren crossover, two dbx 160X compressor-limiters, and two Klark-Teknik DN27 graphic equalizers. Additionally, an Inovonics Model 500 Acoustic Analyzer (fed with an AKG condenser microphone) was handy (Figure 7).

"Consistency is probably my greatest watchword," Pope remarks. "I try to have the gear set up in the same way every night. I EQ the system every single time with the same taped music; that way, I'm used to hearing the same thing in different rooms."

A pair of Revox A77 two-tracks were at hand to provide playback music (Pope's personal selection for setting the

Figure 7: The system rack houses Brooke Siren Systems crossovers, dbx Model 160X compressor-limiters, Klark-Teknik DN27 graphics, and an Inovonics Model 500 Analyzer.
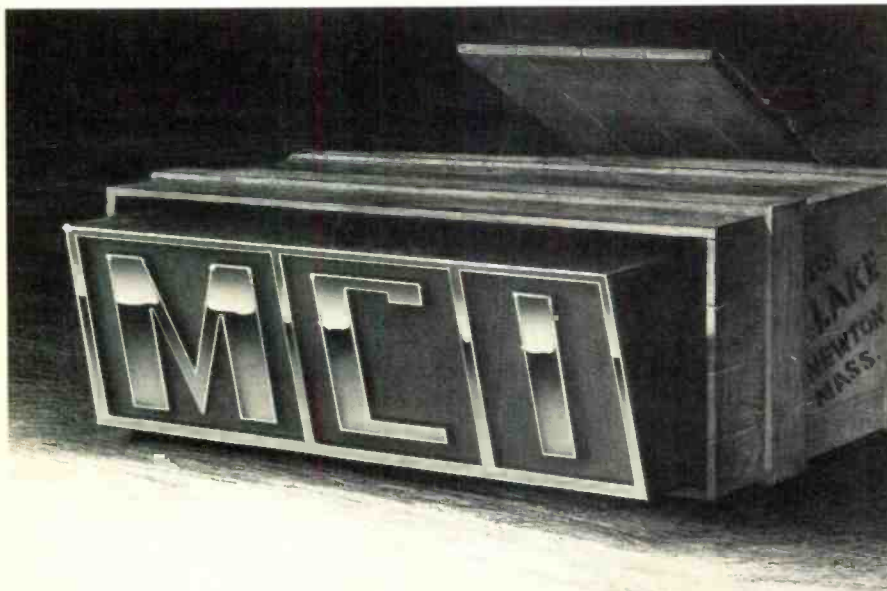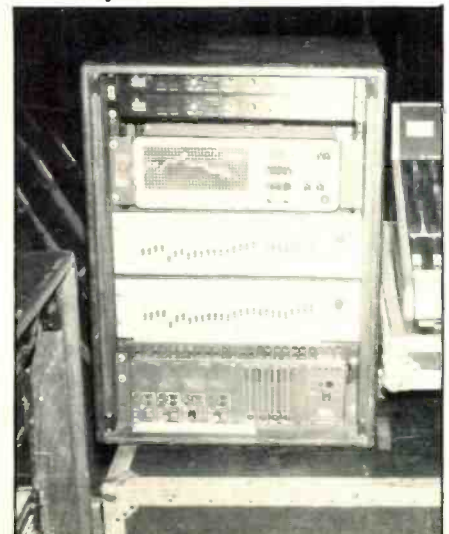
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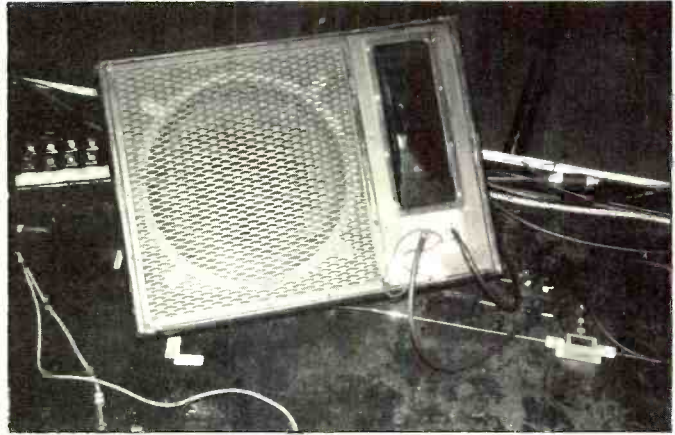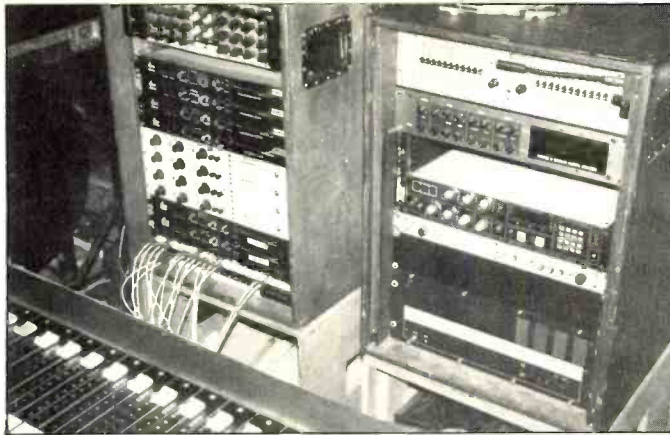
Figure 8: A secondary rack, located behind the mix position, contains a wide selection of signal processors and effects units.

Figure 9: On-stage monitoring was covered with 24 Martin LE200 wedges.

system up included tracks from Pink Floyd's *The Wall*, and Toto's latest release). The two-tracks also provided playback of a film soundtrack, used at several points during the show to accompany projected film clips. Duplicate copies of the soundtrack were loaded onto the two identical tape decks, so that a spare was on-line instantly if needed.

Approximately eight or 10 feet behind Pope, a pair of secondary racks were positioned, which contained all signal-processing devices for channel inserts and special effects (Figure 8). One rack contained four Aphex CX-1 noise gates (for toms, kick and snare), and four dbx Model 160X limiters (keyboards and bass guitar). Additionally, four Inovonics Model 201 limiters were available for vocals. Two more dbx 160s handled the guitar inputs. A full patchbay was provided on the front panel for access to each device.

The second effects rack housed an Aphex II Studio Aural Exciter, an AMS DMX-15-80S stereo digital delay, a Delta-Lab DL-4 Time Line, and a Lexicon 224X digital reverb.

"I use the Aphex as an effect, with selected inputs such as vocals and saxophones," Pope explains. "I use it very sparingly, and not every night; only if the hall seems to need the added help on the top end. The primary thing that we are all into is to remain as totally *musical* as possible, and to have the show sound be as much like the recordings as we can make it. So, the signal processing is here to be used as needed for certain tunes — certain sound textures, if you will — not to be just thrown on everything indiscriminately depending on my mood that show."

"I have a lot more gear out here than I did, say, five or six years ago. But I have always tried to keep it as simple as possible. It seems to me that the more devices you stick into your signal path, the more cluttered everything becomes. New gear is very faddish, sometimes . . . I don't add anything unless I really feel it is necessary.

"And, basically, my system here is an antique, compared to many of the new high-tech rigs out right now. Things like

subwoofers, for instance. Back four or five years ago when that was the new thing to have, we had just finished seven or eight straight years of touring, recording, touring, and we just didn't really want to jump into a whole new thing like that with our system. Actually, now I am glad we didn't — the Martin bass bins are very adequate for low-end. Take this place for instance; the bass in the room is just horribly boomy. It can just make the whole mix turn into jelly, washing out the vocals, the piano, everything. More low-frequency energy is the *last* thing I need."

Pope feels that the system has actually benefited from the current plan

whereby the sound crew has taken over the PA and formed The Delicate Production Company. "Since they have been serving other clients besides us, the guys have really had to upgrade the system, and make it more contemporary. so, it has been good for all of us. The several years off for our group from touring has given the fellows a chance to really beef up the sound system, and it is working better than ever now. As far as hardware goes, I have everything I used to have, and then some more."

Sound Checks

"Every single show," Pope emphas-

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supertramp

izes, "there is *always* a sound check, usually [with] the whole band. Some of them are always here ahead of time to play the room, so we can get it set correctly. I spend most of my time with the low-end, probably — the bass is very important to us. It has to be full, relatively powerful and tight, and the vocals have to be clean and on top. Everything else is pretty much standard.

"Actually, if you were drawing a diagram of our sound, the drums and bass would be the bottom of the pyramid, and the vocals would be the apex. But the drum and vocal sound by itself has about the equivalent acoustical energy as all the rest of the instruments. If you were to measure it out here in dBs, I think that you'd get, say, 110 dB overall level. And then, if you pulled out the mid-range instruments, and left only drums and vocals, you'd *still* get essentially the same registration on your meter.

"If you can establish a solid, clean bottom end, and have the vocals clear and on top, you have to mix with a feeling of power and size. *That* is where many other mixers make the mistake, I think, [of] trying to focus *too* much on guitars, pianos, that sort of thing. An

MARTIN RS1200 FOUR-WAY COMPOSITE CABINET

For the Canadian leg of the Supertramp North American tour, Delicate Productions provided much of the house speaker system, while Audio Concepts A.C. Inc., a Quebec-based sound reinforcement company, was subcontracted to supply a Martin speaker system for many of the shows. Audio Concepts stocks the standard Martin speaker systems, along with Dave Martin's newest product: the RS1200.

"For the indoor arena dates which required supplementary gear, we provided the RS1200s for rearfill," explains Audio Concepts' sales representative Daniel Angers. "Russell Pope [Supertramp's house engineer] commented that he was very pleased with this new cabinet. We hung eight in the air and placed four on the floor at the back of the five largest arenas on the tour. For the outdoor dates, we brought in Martin cabinets identical to those used by Delicate."

The RS1200 was several years in the making by Martin Audio, Ltd. of London, and was only first introduced into North America in July of 1983. "It is our understanding that these boxes are for sale only in large quantities to sound reinforcement companies, and will not be available to the public," Angers states. "Ours were shipped over from England unloaded and unpainted. We installed the components, finished them with a fiberglass exterior, and then installed special steel load-bearing handles for hanging.

"The box has a rather complex loading system; it is horn-loaded in the bass section, somewhat like a pyramid. We use Gauss drivers which are specially-made for us, a 300 watt ATC 12-inch cone, one JBL 2445 driver, and two Fostex aluminum-diaphragm tweeters."

Audio Concepts currently has 32 of the new cabinets as part of its regular sound rental stock. "So far as we know, this is the only RS1200 system in North America to date," Angers comments. "We feel it is an excellent box. It retains the solid, horn-loaded bass and mid-bass that Martin systems are known for, yet packages all the components for a 4-way system into one self-contained cabinet." □□□

electric-guitar-oriented band is very dependent on the guitar riffs. But Supertramp's music is just the opposite. And it just happens to translate well to the acoustics of an arena such as this one. I don't think that the group has ever consciously structured their music to fit room acoustics, but they have always gone for what *they* wanted to hear — and lyrics and melody are most important.

"As far as reproducing the tunes out here on the road, I used to only have a single spring reverb to bring into the mix. The band's music is structured in a very simple manner, and I try to approach the live recreation of it in the same manner. My Midas console was expressly designed to be almost childishly simple. How anyone can put parametric EQ on every channel and expect to keep track of it all during a show ... forget it! The simpler, the better. Technology won't give you better ears, or solve problems in the music itself."

During the show, Pope's attention was focused almost single-mindedly on the Midas main board; drums and percussion submixes from the Trident submixer appeared on inputs routed to the Midas console. The Lexicon 224X's remote control panel also was close at hand; all other effects were pre-set and brought in as needed. As the show progressed, Pope played the Midas as if it were an instrument itself, his eyes rarely leaving the stage. Norman Hall, an engineer with Supertramp, was present to keep a watch on the system drive levels, and the input overload lights on the various effects units.

Monitor System
The Supertramp monitoring system on-stage consisted entirely of Martin LE200 wedges (Figure 9). Twenty of the cabinets were positioned on the large 64-foot wide stage area, with an additional pair being set up at the monitor board for listening to each mix. Output mixes were set up in 'zones' (organ position, piano position, center vocal mike, etc.). Thirteen mixes were fed to the stage, and another two were used as inputs to Lexicon 224 and Delta-Lab DL4 digital delay units used exclusively in the monitors.

"I really don't need sidefills at all," explains Ian Lloyd Bisley. "So much low-end carries on-stage from the house system, that sidefills would only confuse the band. Most people use them, I think, to try to give the performers more presence on stage, particularly in the bass department. The wedges have always been quite satisfactory."

Bisley's console was a new 40-input/16-output Soundcraft Series Four (Figure 10). Additionally, a Yamaha M-1532 was utilized as a submixer to bring up the drums and percussion; its various outputs were then fed into the Soundcraft as several discrete submixes. From the Soundcraft, the output signals went through 13 Klark-Teknik DN27 graphic
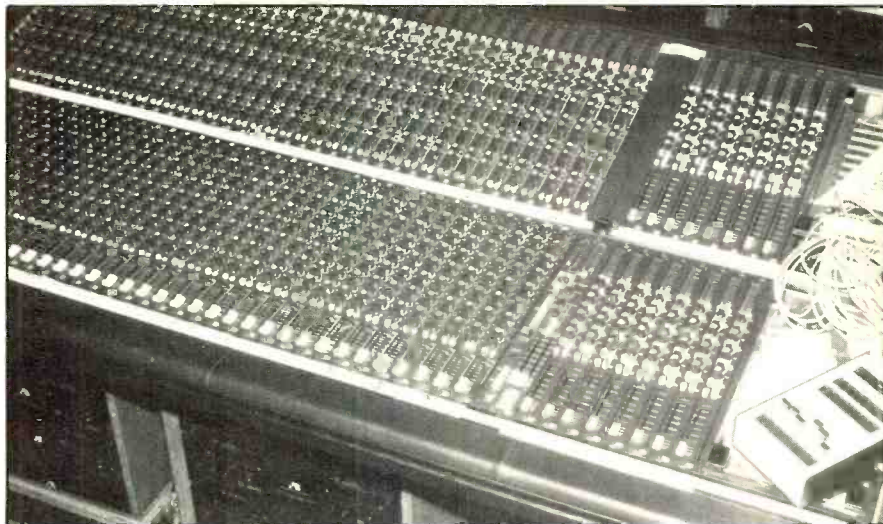
Figure 10: Soundcraft Series Four monitor console, equipped with 40 inputs routing to 16 submix outputs.

equalizers, before hitting 32 sides of AB Model 1210 stereo power amplifiers. Additionally, three sides of dbx limiting were available, which Bisley used on kick drum and Wurlitzer electric piano. Brooke-Siren crossovers provided bi-amplification outputs.

"Because this band does not jump around a lot," Bisley offers, "the live sound has always been a large part of what was important to us. I have always felt that my constant attention to the mixes — to what the guys were

actually hearing — was a lot more important than adding a lot of hardware to my system. Plus, we never really had a lot of extra money to throw around into new gear. So, I learned early on how to make the guys happy with the monitors.

"It has only been recently that we have been able to add things like the equalizers, and the Soundcraft console. I realized after looking at this upcoming tour that I would need the new board. My last board had 34 inputs, but only

eight mixes. I liked the Soundcraft board, after I saw one of the first ones in use on the Styx tour." [See *R-e/p* June 1983 issue — *Ed.*]

This tour, Supertramp had added two additional sidemen to the band, and the stage setup was particularly widespread. "We have more instruments, more inputs, and more output mixes," Bisley continues. "I did have to expand the system this time out. But basically I have always felt that *less was better*; the fewer things in the signal path the better I like it. There's just *no* point in going over the top with your hardware. As far as the graphic equalizers go, I don't like to rely on them too much. After a while, the show just settles down to a norm, and I get to know what frequencies are going to try to feedback during the show. An attentive monitor engineer can do a lot with a very simple system."

Stage Miking

Bisley was taking 46 inputs from the stage into his two consoles. A majority of the inputs were picked up with microphones, while some of the electric keyboards were taken direct. On the drum kit, a Sennheiser MD421 was placed on

TABLE 1: MONITOR CONSOLE INPUTS

Soundcraft Series Four Main Mixer:

1. Talkback from House Mixer
2. Spare
3. Bass guitar DI
4. Kick drum
5. Snare drum
6. Hi-hat
7. Drum submix from Yamaha sub-console
8. ARP Omni synthesizer
9. Clavinet
10. Hammond B-3 organ (Leslie miked; Y-ed)
11. Oberheim synthesizer
12. Grand Piano pickup
13. Wurlitzer; stage left
14. Wurlitzer; stage right
15. Saxophone (Scott)
16. Clarinet (John)
17. Saxophone (John)
18. Wireless Sax Mike
19. Oberheim OBX synthesizer
20. Elka string synthesizer
21. Oberheim OB4 synthesizer
22. Electric guitar (Fred)
23. Electric guitar (Scott)
24. Electric guitar (Roger)
25. Acoustic guitar (Roger)
26. Vocal (Roger)
27. Roland Jupiter 8 synthesizer
28. Elka synthesizer (Roger)
29. Organ vocal
30. Piano vocal
31. Wurlitzer vocal
32. Sax vocal
33. Strings vocal
34. Spare vocal
35. Scott vocal
36. Fred vocal
37. Soundtrack tape input from House Mixer
38. DDL return
39. Lexicon 224 return; left
40. Lexicon 224 return; right

Yamaha M1532 Sub-console:

1. Percussion
2. Tom 4
3. Tom 3
4. Tom 2
5. Tom 1
6. Timbales left
7. Timbales right
8. Overhead left
9. Overhead right

the kick, AKG C460Bs on snare and hi-hat, MD421s on toms and guitar amplifiers. Vocal mikes were Shure SM-85s, including one SM-85 capsule attached to an HME wireless transmitter, used in the center downstage position by several different band members. One of two saxophones also was passed through an HME wireless unit. The grand piano, a critical part of Supertramp's sound, received a Helpinstill pickup equipped

with a custom-built mixer.

Other inputs to the monitor board included talkback from the house, effects tape feeds from the front-of-house mixer, and DDL and Lexicon 224 return lines. A spare vocal microphone also was provided (see Table 1).

Keyboards used during the show, each on its own console input, included grand piano, a Hammond B-3 organ, Roland Jupiter 8, Oberheim and Elka

Full outdoor Martin system in use during Supertramp's European Tour, Barcelona, Spain.

synthesizers, and two Wurlitzer electric pianos.

Sound Of The Show

Now, right here, I have a confession to make. I almost did not want to review this system, as Supertramp has long been a favorite of mine on record, and I did not want to be disappointed by poor live sound. For the record: I was not disappointed.

The Checkerdome in St. Louis is a large, oval sports facility with a wooden roof. Draperies have been hung up above the stage to try and counter the reverberant tendencies of the hall, and many sound systems have tried unsuccessfully to do battle with the cavernous venue's boomy, bass-heavy sound. Before this show, I was a bit skeptical of the seemingly small system. I needn't have worried. Walking throughout the entire venue, I noticed that the sound actually improved as I went higher into the seating areas. At the extreme rear of the hall, in an area known as the Upper Box, I could actually understand most words which were sung, even on unfamiliar tunes.

The sound of the system was quite warm; it lacked the almost artificial super-high brilliance of some contemporary sound systems. There was a tremendously powerful bass/drums underpinning structure to the mix, true to Russell Pope's pre-show philosophizing. Yet, the bass frequencies did not overtake the rest of the mix. At no time did I detect what I would call "boominess" —yet, there was still a very strong, "present" feeling to the kick drum and bass guitar, and synthesizer lines.

Dave Martin's ideas on low-frequency reproduction did, indeed, pan out. The faithful sound engineers for a once-unknown English band got their wish: a great PA to work with. The road crew got their wish: their very own sound and lighting company. And the band? "We never really had any doubt that, eventually, things would get to this point," stated bassist Dougie Thomson as he surveyed the sold-out arena. "We knew it was possible. We just went about it in the best way that we knew how."

Fairy tale, no. Success story? Absolutely. ■■■

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RETURN OF THE JEDI

Sound Design for the "Star Wars" Trilogy by Ben Burtt
Part Two: Final Dubbing Process

by Larry Blake

Part one of this article, published in the October issue, described how Lucasfilm sound designer Ben Burtt and his crew gathered the necessary production tracks in the studio and on location, and then created the various sound effects. In this, the conclusion, we will see how the composite sound elements came together at Sprocket Systems' dubbing stage, and the particular emphasis placed on dialog and effects pre-mixes for the final 35mm Dolby Stereo optical and 70mm six-track dubbing process.

Final Stereo Film Mix

The dubbing of a stereo motion picture presents a mind-boggling number of monitoring and recording variables to re-recording mixers. Among the questions that have to be answered: Discrete six-track (five full-range speakers) versus Dolby "boom" six-track; surround and boom tracks (yes or no? how loud and how important? What information?); Dynamic range (how loud is the monitor level? relationship between dialog and 100% level on the track); Discrete four-track monitoring versus monitoring through the Dolby Stereo matrix; Mono compatibility of Dolby two-track master; and so on.

Having only been involved with stereo films, Ben Burtt had given these matters much thought, and the mix of *Return of the Jedi* was his first opportunity to apply them as a re-recording mixer. However, until the latter stages of the mixing process Burtt remained in his mix studio doing effects pre-mixes, while his long-time assistant, Gary Summers, worked in the dubbing stage, handling the dialog and the other effects pre-mixes.

Joining Summers, for whom *Jedi* also was his first dub, was Roger Savage, an Australian whose best-known work in the U.S. is *The Road Warrior (nee: Mad Max II)*. Burtt and Summers met Savage while he was visiting the Sprocket Systems complex in Marin County, north of San Francisco, and a week later offered him the chance to work with them. Savage was on hand during effects and dialog pre-dubs, and for four reels of the finals, when he handled the music. After that, Randy Thom looked after the music for the rest of the film mix.

The first job for Summers and Savage was the dialog pre-mix. In a *Star Wars* film this means, in addition to the usual tasks of matching looped lines with production tracks, and cleaning up background noise whenever possible, applying the signature processing of C-3PO and Darth Vader. Summers found out that if there was too much going on in a particular scene, occasionally he would have to go back to the original (non pre-mixed) track for C-3PO or Darth Vader, and remove some of the processing to make the voice more intelligible. Similarly, in a few instances he had to go back to the original elements, and re-mix C-3PO with more processing if there was nothing much else happening on the screen.

"No reverberation processing was done in the dialog pre-mix," Summers notes. "It was all left to the final mix, so we could judge how much reverb to put on each sound."

Summers' main tool in reverb processing was a Lexicon 224-X, whose split programs he found to be extremely useful. "When you split the 224-X it really becomes *two* discrete reverb units," he says. "This way I was able to send Jabba the Hutt, Salacious Crumb, and so on, to one side, and that return was bussed to the 'creatures' track. The other side was used for English dialog. In addition, I was able to change the parameters of the Lexicon for Jabba versus Luke while they were speaking."

Even though the English shoot took place exclusively on soundstages, Summers notes that "some of that I would have preferred to have had looped but, as it turned out, it wasn't. A lot of times they selected the production sound over looping, because the performance was better.

"The most important factor in getting good production sound has to do with cooperation on the set. Will the director or the cinematographer give you adequate time to prepare . . . to keep the people quiet?" (After the mix was finished, dialog coordinator Laurel Ladevich went through the picture and tallied up all of the looped lines, versus production tracks, and found that 83% of the picture had been dubbed.)

"This was the worst picture I've ever worked on in terms of changes," notes Burtt. "We were going back and starting reels over for a third or fourth time." After what was considered a final 12-track mix had been finished, there were changes in 11 of 13 reels in the film, and then the crew had only 10 days to finish the final mix before George Lucas had to leave for Sri Lanka, and the start of shooting for *Indiana Jones and the Temple of Doom*.

Separating Dialog and Multiple Effects Tracks

Although punch-in (a.k.a.: reversible update; insert; rock and roll) recording first appeared on the film scene in the early Sixties, its use in stereo mixes was not widespread until the mid-to-late Seventies. Along the same lines, among the major studios only MGM historically has recorded separate stereo dialog, music, and effects (DME) elements as the master mix for a stereo film, thus facilitating the necessary separation of dialog from music and effects to create foreign stereo printing masters. The

RETURN OF THE JEDI

addition of punch-in recording made the initial mix less of a chore, since the music and effects mixers didn't have to re-do their parts if, for instance, the dialog mixer made a mistake during the final mix.

All other studios that dubbed stereo films had to make separate mixes to extract DME mono and foreign stereo elements. Even today, only a few studios — for example, MGM, Lion's Gate, Todd-AO, Warner Hollywood, and The Burbank Studios — record separate stereo elements during final mixes. This is a somewhat ironic note since, with the coming of magnetic sound, the film industry immediately saw the benefit of separating dialog, music, and effects elements during less complicated mono mixes.

On the three Star Wars films, Ben Burtt has worked with three different, increasingly flexible stereo mixing procedures. Star Wars was mixed to a composite four-track master, which caused Burtt many headaches. "When you have to do a separate mix for foreign M and E stereo, you don't do as good of a job [as you did on the initial stereo mix] because you're pretty burned out at that point. You don't care anymore. You try to remember what you did before, and you struggle with exactly the same problems," he says.

"Around the time of Star Wars, they were afraid to punch in on stereo: 'You just couldn't do it.' There would be a major discussion for a half hour in the hall: 'Can we punch in?' 'I don't know.' They would rather do a whole reel over than punch in. People weren't comfortable with stereo."

For The Empire Strikes Back, four-track composite and four-track effects-only elements were recorded. This only helped take the headache out of making foreign stereo printing masters, since only the music, consisting of two, three-track elements, needed to be re-mixed. Nevertheless, since everything was tied together, all units needed to be put up,

and everyone had to match levels to make the slightest change in the four-track composite.

On Return of the Jedi, Ben Burtt and Gary Summers took the idea of recording separate dialog, music, and effects strips one step further, and split the effects into "creatures," and "everything-but-creatures." Two six-track recorders were pulled for the master mix, recording left-center-right mixes of English and creature "dialog" on one, and of effects and music on the other. No surround information was recorded on the master mix.

The "creature" mix contained R2-D2, Darth Vader's breathing, Jabba the Hut, Salacious Crumb (Jabba's obnoxious, giggling sidekick), Bibb Fortuna, Nien Nubb, Boussh (the bounty hunter whom Princess Leia is disguised as), the eye robot that greets visitors at Jabba's palace and, of course, the Ewoks. What effects were left — laser swords, spaceships, backgrounds, and garden-variety Foley — were recorded on the effects

— Gary Summers —

"split." The movements of R2-D2 and C-3PO were mixed to the Foley pre-dub, and later bussed to the effects split during the final mix.

The reason for separating into two areas all non-music and English dialog elements is clear, in that either one of them — creatures or standard effects, as used in Jedi — is as complex as all of the effects on a standard stereo film.

"When you're tying everything together, you don't have the flexibility you need to make changes," Burtt notes. "Especially in our type of movie that's being recut until the very last moment. You're constantly remixing — not necessarily because there was anything wrong with the sound, but often because they put two or more spaceships in the shot that were never there before, or they've taken two spaceships out, and you've got to keep everything the same, but get rid of those two ships. You don't want to remix everything else totally for that shot. You just go in and change the effects track. The creative flexibility is so much greater."

— Ben Burtt —

— Roger Savage —

This splitting of stereo information, which many consider to be the rough equivalent of automation, coupled with the presence of Sprocket System's Neve NECAM II automation system, possibly makes Jedi the first film to use both techniques. Burtt notes that the dubbing engineers had a "pleasant relationship with the NECAM, using it for the things that we knew it would do best for us, and avoiding it when it was simpler to go manual." Also, the NECAM automation software was written for two-minute record mixes, and not for a complex 10-minute film dub.

Gary Summers elaborates: "I didn't like having to eventually merge many small [mix] updates that I had made in the course of doing a mix, and wish that all updates would be recorded on to one master mix."

Nevertheless, Burtt notes that "the thing about NECAM that works so great for film mixing, because effectively it's a mechanical system [via servo-controlled faders, instead of VCAs], is that a mixer wants to watch the screen and listen. You're not looking at the fader. Many of the automation systems require that you match two little lights, and pretty soon you're mixing the data, not the movie. All your attention is on, 'Did I punch into the data okay?' It's like having two different mixes at the same time. You want to have an automation system that does what your hand wants to do next."

Although the Neve 8108 console at Sprocket Systems has 32 main inputs (plus the 24 monitor faders if they are used), Burtt and Summers never felt a need for any more channels because, in effect, they had worked backwards from this number, and designed the track layout to fit the board configuration. In addition, all of Burtt's previous films — Star Wars, More American Graffiti, The Empire Strikes Back, and Raiders of the Lost Ark — had been mixed at Room D at Warner Hollywood which, at that time, had a 33-input Quad-Eight board.

The console layout during the final mix was almost identical for every reel: six tracks of dialog — three English, and three creatures; two, three-track
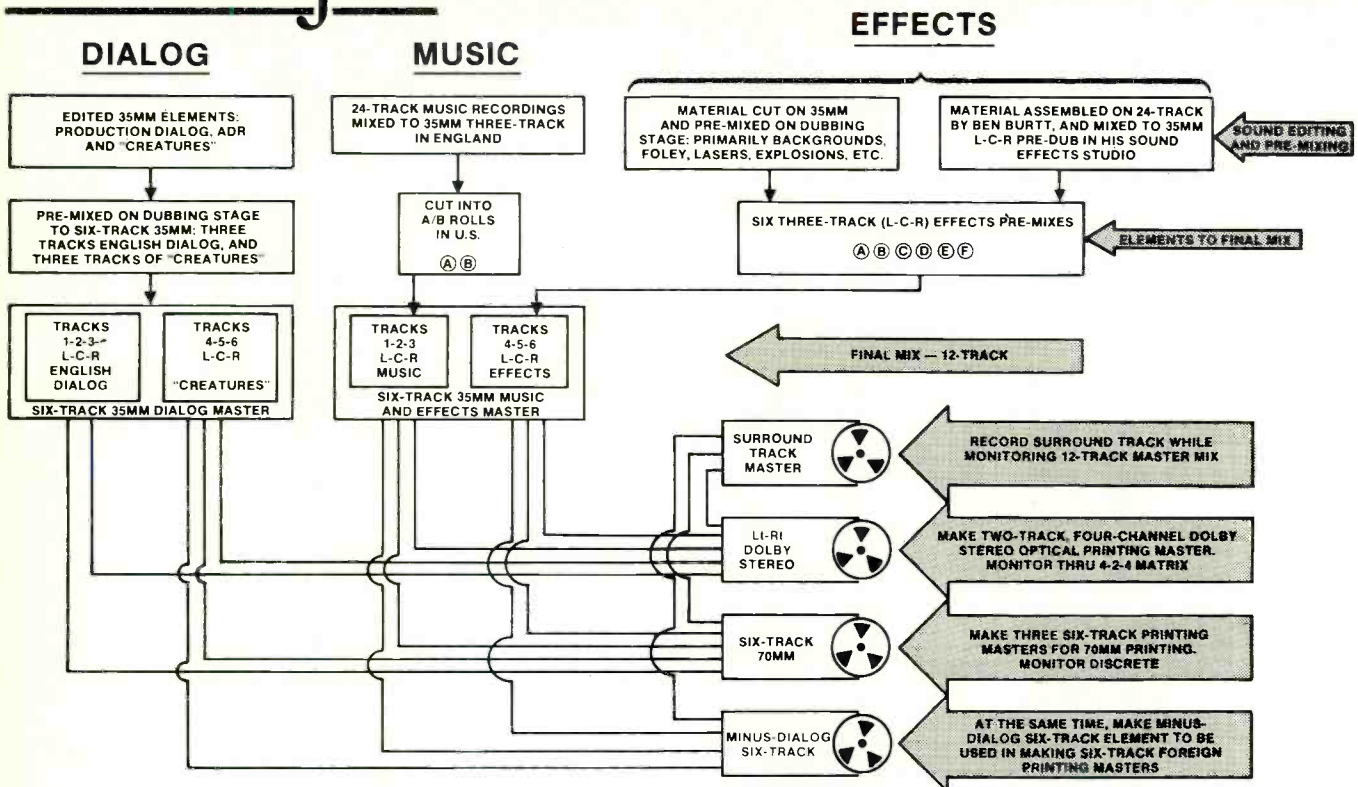
FLOW CHART OF SOUND ELEMENTS FROM EDITING TO PRINT MASTERING

DIALOG

- EDITED 35MM ELEMENTS: PRODUCTION DIALOG, ADR AND "CREATURES"
- PRE-MIXED ON DUBBING STAGE TO SIX-TRACK 35MM: THREE TRACKS ENGLISH DIALOG, AND THREE TRACKS OF "CREATURES"
- TRACKS 1-2-3 L-C-R ENGLISH DIALOG
- TRACKS 4-5-6 L-C-R "CREATURES"
- SIX-TRACK 35MM DIALOG MASTER

MUSIC

- 24-TRACK MUSIC RECORDINGS MIXED TO 35MM THREE-TRACK IN ENGLAND
- CUT INTO A/B ROLLS IN U.S. Ⓐ Ⓑ
- TRACKS 1-2-3 L-C-R MUSIC
- TRACKS 4-5-6 L-C-R EFFECTS
- SIX-TRACK 35MM MUSIC AND EFFECTS MASTER

EFFECTS

- MATERIAL CUT ON 35MM AND PRE-MIXED ON DUBBING STAGE: PRIMARILY BACKGROUNDS, FOLEY, LASERS, EXPLOSIONS, ETC.
- MATERIAL ASSEMBLED ON 24-TRACK BY BEN BURTT, AND MIXED TO 35MM L-C-R PRE-DUB IN HIS SOUND EFFECTS STUDIO
- SOUND EDITING AND PRE-MIXING
- SIX THREE-TRACK (L-C-R) EFFECTS PRE-MIXES Ⓐ Ⓑ Ⓒ Ⓓ Ⓔ Ⓕ
- ELEMENTS TO FINAL MIX

FINAL MIX — 12-TRACK

- SURROUND TRACK MASTER — RECORD SURROUND TRACK WHILE MONITORING 12-TRACK MASTER MIX
- LI-RI DOLBY STEREO — MAKE TWO-TRACK, FOUR-CHANNEL DOLBY STEREO OPTICAL PRINTING MASTER. MONITOR THRU 4-2-4 MATRIX
- SIX-TRACK 70MM — MAKE THREE SIX-TRACK PRINTING MASTERS FOR 70MM PRINTING. MONITOR DISCRETE
- MINUS-DIALOG SIX-TRACK — AT THE SAME TIME, MAKE MINUS-DIALOG SIX-TRACK ELEMENT TO BE USED IN MAKING SIX-TRACK FOREIGN PRINTING MASTERS

music reels; and six, three-track effects pre-dub reels, including Foley plus A-B-C-D-E effects, ranging from backgrounds to more important foreground effects. Some of these effects units were pre-dubs that Ben Burtt had made in his studio, by remixing from the 24-track. What he didn't have time to do was handled by Summers and Savage in the dubbing theater.

Surround and Boom Channels

As indicated, surround information was not recorded during the initial 12-track mix. Once that mix was locked, and approved by George Lucas, a separate surround track was recorded on another piece of fullcoat 35mm mag, while monitoring the behind-the-screen mix. All of this surround material (track #13 of the master mix) was cut specifically for the surrounds; no material was taken from a pre-dub up front.

This procedure, which Burtt first used on *Empire*, in reality dates to a lesson he learned by hearing *More American Graffiti* in the real world outside of the dubbing stage. "During the mix of *More American Graffiti*," he recalls, "we put tons of stuff in the surrounds all the time during the mix, and later found out that most of it was never played correctly in theaters. In some instances we had music in the surrounds at a significant level, and if you took the surrounds away it was *gone* up front. We learned a hard lesson on that.

"My attitude [on *Jedi*] was: the sur-

rounds really don't exist — nothing was ever destined for the surrounds that was considered an essential piece of information. We went back and did the surrounds with the front already done, as if that were the final movie. We *didn't* depend on the surrounds for any principal dramatic effect because the surrounds are the least likely to be played correctly. The surrounds are an *enhancement* to what is already there, and we trust that if the film is played under something resembling standard conditions it will work fine.

"If I had a spaceship pass-by from back to front, I would have a complete pass-by in the front. In the surrounds would be a pre-pass-by — a sound that would come right before the ship entered the frame — so that if it didn't get played, all you saw was the ship fly by. If it *did* get played, you would hear it coming for an instance, and then it would go to the front. In some cases it gave you a nice effect of coming over the audience's heads. [But] it was not derived from the sound up front." Sometimes Burtt would make a mono mix of the music that also would be layed off for the surround mix.

While this surround information, and only this surround information, would be heard on the 70mm prints, the Dolby Stereo optical matrix will bring to the back of the theater *any* information, that is recorded out-of-phase between the two tracks on the 35mm print. Ben Burtt refers to the information that the

Dolby Cat No. 150 matrix extracts as the "magic surrounds."

"There are certain wonderful things about that in some cases," he says, "because basically the information that gets into these magic surrounds is non-directional, out-of-phase material. It tends to enhance some things in a very pleasant way. For example, in certain crowd scenes we had the Ewoks running around, panned left and right around the front. In the matrix they were bent around into the surrounds, and it was a much more spatial effect than I got in the 70s [70mm prints]. It sounded more realistic, and was an *accident*.

"We might consider putting a send on our board from the matrix, so we could record what the magic surrounds were doing at any moment, and use it in the 70mm six-track version."

Just how much information the matrix pulls into the surrounds became evident when the TAP "800" line [see THX sidebar — *Ed.*] started getting calls from projectionists in theaters that initially played *Jedi* in 35mm, because a 70mm print wasn't yet ready.

"In a few cases we got calls, 'What happened to the surrounds? They're not here anymore.' In 35 [Dolby Stereo] optical there is so much material there anyway, especially the music and its out-of-phase nature, that a certain 'blob' of it is always in the surrounds [with 35mm stereo optical] that is not there in discrete 70mm." (*Apocalypse Now*, whose six-track mix pushed surrounds

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RETURN OF THE JEDI

to their technical and artistic limits, ironically was released in 35mm Dolby Stereo without intentional surround information. Walter Murch, sound designer for *Apocalypse Now*, shares Ben Burtt's fear of worst-case playback, and felt that the potential loss in such situations was greater than the gain would be in an average theater. Great care was taken in the 70mm installations to make sure that the surrounds were being played properly.)

The boom channels also are regarded by Burtt as an enhancement, and not as a primary creative tool. *Star Wars*, the very first Dolby "baby boom" film, had the boom channels switched on throughout the whole film, with *all* information — dialog, music, and effects — below 200 Hz from the left and center, and center and right, fed to channels #2 and #4, respectively. This low-passed material was on throughout the whole film, and occasionally the material was "spiked" 6 dB for explosions, etc.

Since then, standard industry practice has evolved to where the boom channels are used selectively, usually only in big effects sequences. Placement of music in the boom channels varies; some mixers never use them, some use them all the time.

"Through *Star Wars, Empire,* and *Raiders,*" Burtt says, "my basic attitude about the boom has come to where I'm using it less and less. You're always in trouble using the boom, to a certain extent, because you don't know how the average theater is going to handle it.

"Often we would show up unannounced at a theater, and ask if we could run our film without first checking the system. We took one of our first [six-track] mixes to the Coronet [in San Francisco], played it, and the boom was so loud, because they had subwoofers, that the chains on the outside exit doors were slapping. Things were shaking, and it was *disturbingly* loud. We took the same mix to the Northpoint, and it was like, 'I'm not sure I hear it.' You *have* to play it safe; use it sparingly, and come up with a compromise.

"This is especially true in a film like ours. There are many spots in the film that could justify the use of loud low-frequency material, but we reserve the use of it for those moments when we want to catch the audience by surprise."

Problems of Dynamic Range

When a temp mix was played at an early screening, music that was at a low level relative to the dialog disappeared beneath the audience noise, which in the motion picture industry is referred to as "popcorn noise." Lucasfilm chief engineer Tom Holman made some acoustic measurements, and determined that Sprocket Systems' dubbing

room has 20 dB more dynamic range on the bottom end (soft) and 6 dB on the top (loud) than an average, good theater; these measurements assumed that patrons are not present, but that air conditioning is on.

It should be noted that these problems would have been exacerbated had the monitor level not been carefully kept at the Dolby Stereo standard of 85 dBc/slow, measured at the console with 50% on the meters, a level determined by a study Dolby Laboratories made of the average fader level of first-run theaters.

To effectively reduce the dynamic range of the Sprocket Systems' dubbing stage, three separate, uncorrelated noise sources shaped to make NC-30 noise were built, and placed separately in the left and right surrounds, and in the front. NC-30, not coincidentally, is the maximum value allowed for a THX theater installation [see accompanying sidebar].

After the 13-track master mix was locked, the next step for Summers and Burtt was to make the Lt-Rt (Left-Total/Right-Total) two-track printing master for the over 900 35mm prints needed for the initial domestic release of *Jedi* on May 25, the sixth anniversary of the release of *Star Wars*. This was before the six-track printing masters for 70mm release would be made.

The observant reader might find this timing rather odd, because 70mm prints take longer to make, *per se*, than 35mm prints. The reason the Lt-Rt printing master had to be made first is simple: the optical soundtrack is printed at the same time as the picture, whereas 70mm prints have to be striped, and then "cured" for at least a day, before the six-track printing master can be transferred. Thus, theoretically, the 150 70mm prints could have been printed and striped *before* the six-track mix was finished and ready for transfer; this was not the case in reality, however.

Having to mix the Lt-Rt first was an abrupt change for the mixers, since the 13-track mix had been monitored, for the most part, in the discrete mode. Listening to the mix through the matrix

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— Tom Holman —
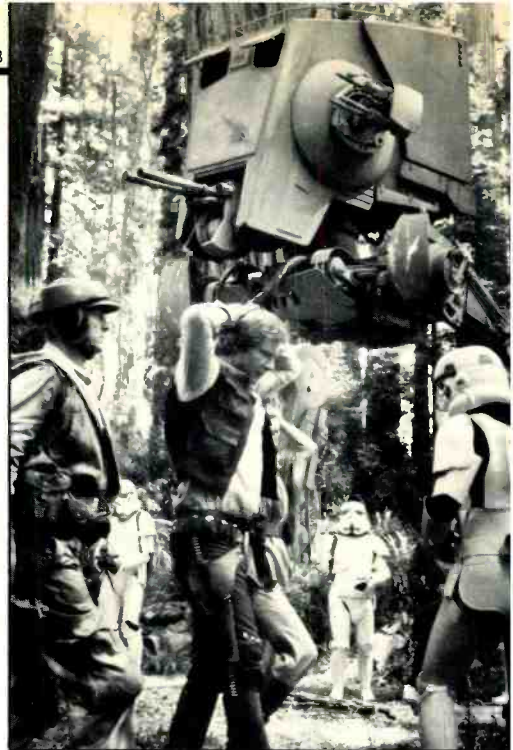
made the mixers aware that the dynamic range would have to be toned down for Dolby Stereo optical release.

Monitoring through the 4-2-4 matrix with the Dolby DS-4 encoding unit effectively simulates what the two-track mix will sound like as an optical print, including the effect of optical "clash." An optical track, like a digital recording, has specific, finite limits that cannot be exceeded. As Tom Holman notes "There's a problem monitoring through the matrix that has to do with headroom. The matrix had adequate headroom for doing an optical track, but the [70mm] magnetic film has considerably more headroom than that. So you don't want to make those adjustments and compromises until later on [when you make an Lt-Rt stereo optical printing master]."

"We kept looking at it as, 'Well, we're going out with 150 70s,'" adds Ben Burtt, "so we tended to gravitate towards going for maximum effect in the 70mm prints, hoping that it would then fit into the 35s.

"The truth of the matter is that we started doing it that way [through the matrix] but our producers always wanted it louder. 'Is the pot up all the way on the music? Is it as loud as you can make it?' We would say that this was as loud as we could record. 'But is it as loud as you can make it?' No.

"So we got off on some things in our initial mix of the film, where the dynamic range was probably too great. In our inexperience as mixers we had some things too loud, and some things too quiet. I think that, with experience, you could mix discrete, knowing how it will work with the matrix."

Another consideration in two-track mastering, in addition to toning down the dynamic range, was how a 35mm stereo print would play in a standard Academy mono projector. When the

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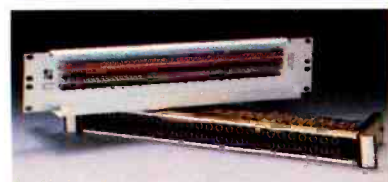
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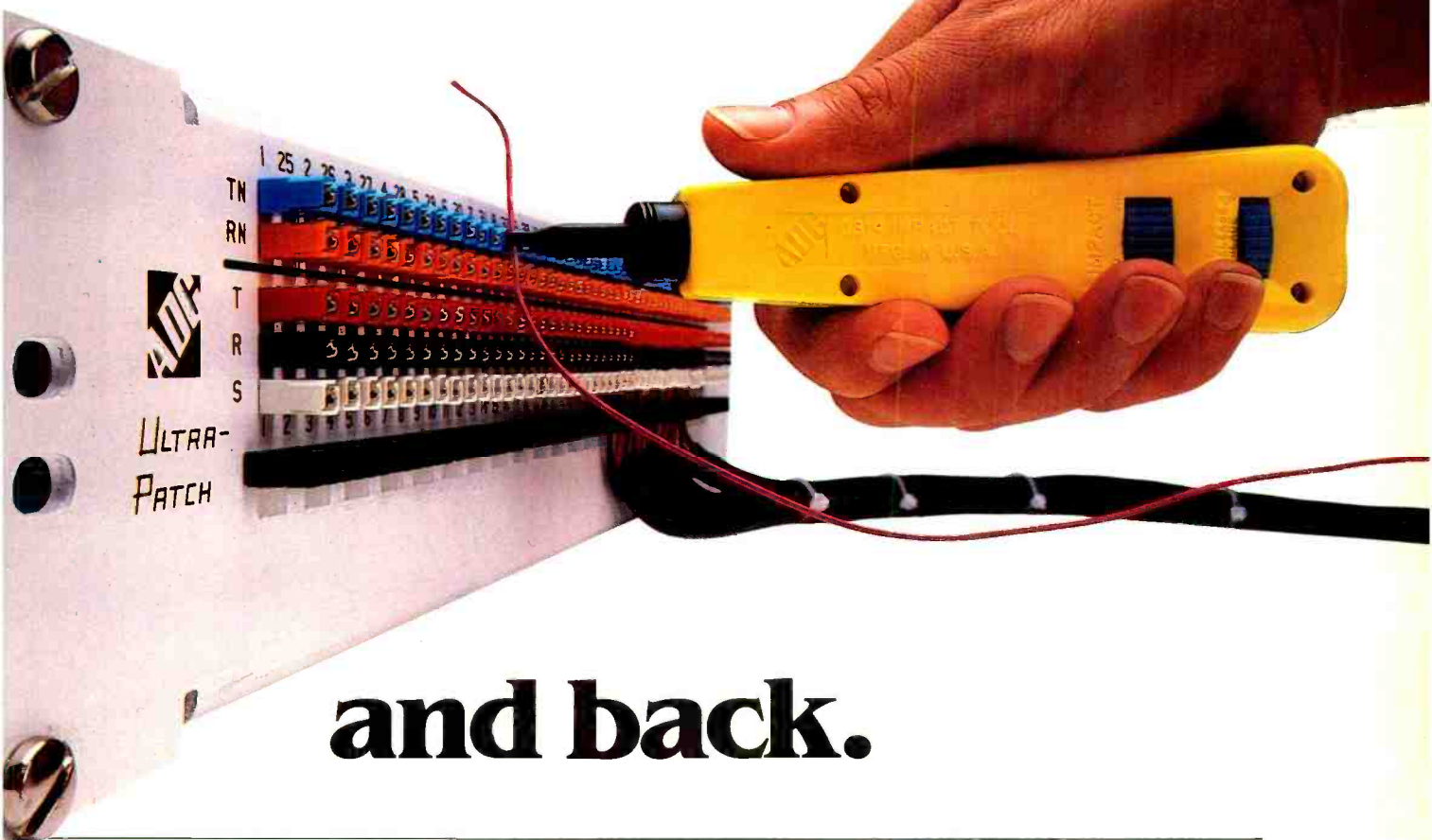
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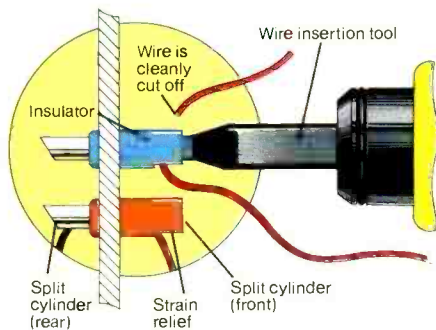
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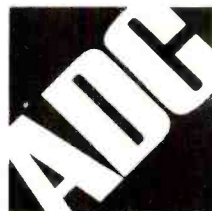
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"mono" button is pressed on the DS-4 unit, the Lt-Rt information is combined into the center speaker, and the monitoring curve goes from wide-range to Academy response. This was a depressing but necessary moment of truth for the mixers of *Jedi*, since the "intergalactic" release in 35mm would be in Dolby Stereo prints only. And, of course, not every theater is yet equipped to play Dolby Stereo prints.

If the Lt-Rt was a compromised version of the discrete mix, having to take into account mono playback further reduces the dynamic range of the stereo mix. Which is not to mention that a Dolby Stereo print played mono is not considered to be the equal of an Academy mono print of the same material. The reason stereo-only release is so popular is that it is so much trouble for the distributors to maintain dual inventory of Dolby Stereo and Academy mono 35mm release prints; single inventory guards against the chance of mono and stereo prints getting mixed up.

After the Lt-Rt had been mixed for a reel, it was monitored all the way through in mono, listening in particular for the areas that didn't fold down to mono acceptably. Ben Burtt: "While we couldn't take care of everything, we took care of things that we objected to; that sounded the worst in mono. For example, in the Sy Snoodles number we had to roll out a lot of low-end because [in mono] all you heard was the beat." Later, a three-track Academy mono mix was made, primarily for use in advertisements and trailers, and also for 16mm release overseas in armed forces bases. In addition, the DME mono mix was, as is standard policy, required contractually.

A little-used feature of the Dolby DS-4 monitoring unit is the bass-sum switch, which combines low-frequency information below 100 Hz into the center channel. This facility only is needed when making the two-track stereo optical printing master, just as similar techniques are used in disk mastering to prevent excessive vertical modulation. Tom Holman explains its use in films: "If you have extreme bass on one channel, it combines the low-end of the two channels so as to improve the headroom of the low bass. [As a result] you could put more low bass on than if you had it totally to one side.

"We had a lot of things in the low-end that we had to roll out more of in the optical, and it ate up a lot of space. These are probably all things that other mixers have learned. A Bill Varney who has mixed three dozen stereo optical films is probably laughing at all of this!"

The next step was to make the six-track printing masters to be used at Twentieth Century-Fox, Warner Hollywood, and Todd-AO in sounding the 150 English-language 70mm prints for release in North America. The 13-track mix, for the most part, could be played straight across. What adjustments were made were duly noted by the NECAM II automation, helping to make three identical first-generation printing masters. Two were made in one pass, and on pass #2 the second recorder captured a minus-dialog six-track mix to be used in making the foreign 70mm printing masters.

El Regreso Del Jedi

The foreign market — that is, everywhere outside of the U.S. and Canada —usually accounts for approximately 40% of the worldwide rentals of a major studio release. (Rentals are the monies returned to the distributor from the theaters, and usually account for about half the money actually received at the box office. This percentage varies widely from picture to picture: The Bond films make 70% of their money from foreign rentals, and only 30% from North America; with the *Superman* series, the figures are reversed.)

The first two *Star Wars* films were closer to the average, as 60% came from domestic, and 40% from foreign receipts. Probably more than any films ever made, these films require meticulous looping and re-recording of the foreign tracks to ensure that the sense and impact of the English-language version truly "translates."

RETURN OF THE JEDI

One of the biggest problems with conventional Earth-bound films is the replacement of sound effects that are lost when the English dialog track is ommitted. This is especially true of "gritty" films, such as *The French Connection*, whose emphasis is on the feel of the original production dialog, and for which production ambience is important. As a result, a relatively large percentage of the aural impact is lost when the English dialog track is removed.

This problem never occurs on the *Star Wars* films, because, as noted, not only is most of the production dialog replaced, but even the simplest background sounds have to be created, and bear no relation to what is on the production track. Prominent among the challenges is the distinctive character of the voices of C-3PO, Darth Vader, and Yoda. (Imagine how differently C-3PO would "feel" if he sounded like a used-car dealer from Brooklyn. In fact, as noted in the book "Skywalking: The Life and Films of George Lucas," by Dale Pollock, this was Lucas' original idea.)

"On the first film we spent a lot of time

— Laurel Ladevich —

researching the foreign character voices," Burtt recalls. "We would audition tapes sent over from the various countries. This time we only had sent over the new characters, like the Emperor and Miff Jerjerrod. The same people have done the revoicing [of the major characters] in each of the three films, and they pretty much know what the film is about."

Burtt accompanied the first *Star Wars* to France, Italy and Spain, supervising the foreign language dubbing and re-recording. Since no stereo music and effects composite was ever made, he had to do a certain amount of remixing every time around, in addition to trying to persuade the mixers to reproduce the dialog, music, and effects balance of the original domestic mix.

Care was taken to avoid these problems on *Empire*, by having the foreign six-track and Lt-Rt printing masters made by the same people (Bill Varney, Steve Maslow, and Gregg Landaker) in the same room — Stage D at Warner Hollywood — as the initial domestic stereo mix. A unique aspect of the foreign dubbing of *Empire* was its being dubbed into Japanese. This is almost never done, due to the large number of English-speaking people in Japan.

Return of the Jedi will be dubbed into six languages: French, standard "Latino" Spanish (for Mexico, South America and the U.S.), Castillian Spanish (for Spain only), German, and Italian. Masters for all of these versions, except the Italian, was handled by Gary Summers at Sprocket Systems. Italian law dictates that no films dubbed into Italian can be imported. In this instance, a six-track composite M and E was sent there, with specific instructions for the processing of voices.

THX THEATER SOUND REPRODUCTION SYSTEM
A Vented-Box, Direct-Radiator Enclosure Featuring a Custom Crossover Designed by Lucasfilms' Tom Holman

To film buffs, THX is the first name of the titular character in both George Lucas' award-winning USC short, *THX:1138:4EB*, and his first feature film, *THX 1138*. This summer THX acquired a new meaning for Lucas, since it now serves as the name of both the trademark that will function as a "Lucasfilm Seal of Approval" for motion picture theaters meeting the company's picture and sound criteria, and of the active electronic crossover that forms the heart of the sound system, designed by Lucasfilm chief engineer, Tom Holman.

Initially, the idea began simply to install a state-of-the-art monitoring system for Lucasfilm's new re-recording stage. Drawing upon the work of Siegfried Linkwitz, the trio of Stanley Lipshitz, John Vanderkooy, and graduate student Peter Schuck, of the University of Waterloo, Canada, designed a passive crossover network to achieve the desired objectives. Gordon Jacobs and Tom Holman then fabricated an equivalent electronic crossover network.

The electronic crossover was designed to complement a set of JBL drivers. The woofer system is a vented box, direct-radiator design, as opposed to the industry standard, horn-loaded reflex design that dates back to 1948, and John Hilliard's pioneering work on the Altec Lansing Voice of the Theater A-4 speaker, and, before that, to his work at MGM in the Thirties on the Shearer Two-Way Horn.

Chief advantage of a horn-loaded reflex cabinet was its efficiency, which represented no small consideration in those days of 3,000-seat theaters, and very few watts per dollar. The design trade-off has been primarily concerned with uneven power distribution and, consequently, a large amount of equalization necessary to meet the wide-range monitoring curve used in the mixing and exhibition of today's Dolby Stereo soundtracks.

By mounting the speaker in a wall, Holman was able to extend the unequalized frequency response down to 40 Hz. "The THX system is good to the 40 Hz third-octave band,

THX sound system at Sprocket Systems' re-recording stage.

SPROCKET SYSTEMS SPECIFY ADC CUSTOM PATCHING SYSTEM

To simplify the complex patching requirements throughout Sprocket Systems' post-production facility, chief engineer Tom Holman specified a custom-designed, pre-wired patchbay system manufactured by ADC, and selected for its high degree of flexibility and reliability — *Editor.*

The dubbing/looping for each language took approximately two weeks, and usually entailed no more than 15 actors. Once the script was translated into the required language from the English "cutting continuity" — a virtually stenographic record of a film, indicating picture and sound cuts to the frame — another person translated it back into English, to effectively double-check. To assist in the voice dubbing, the English DME mono was sent for reference. (An interesting aside: While checking subtitles for *Woodstock*, the filmmakers were aghast to find out that the old lady who had made the original English cutting continuity had translated Sly Stone's "I want to take you higher," to "I want to buy a Honda!")

At the time of writing, only the French and Latino Spanish versions had been prepared. Laurel Ladevich, who edited the dubbed tracks, reports that the only problem encountered had resulted from the difficulty in having the foreign actors playing C-3PO speak Huttese (in Jabba's palace) and Ewokese (most notably in the wonderful scene in which C-3PO recounts the *Star Wars* saga in 25-words-or-less).

We felt we couldn't have C-3PO in Tony Daniels' voice speaking Ewokese," Laurel comments, "and then have someone else's voice speaking French. We had to rely on the foreign actors speaking true Ewokese, and it never really happened. Ben did provide a phonetic breakdown of Ewokese, but the foreign actors didn't want to sit down and learn it on a tight schedule." It was finally found helpful to send the actors a copy of Tony Daniels' readings, so they could have some frame of reference.

All other countries will see *Jedi* subtitled in their native language, although even the dubbed prints will contain a certain amount of subtitling, just as the English prints do, during passages involving both Jabba the Hut and Boussh.

One item that cannot merely be subtitled in the standard manner, but actually has to be re-shot, is the neo-Buck Rogers/Flash Gordon chapter prologue crawl at the beginning of the film. Six different language versions, to match the dubbed versions, were prepared and incorporated into release printing negatives. Also reshot is the scene-setting, "A long time ago, in a galaxy far, far away . . ." card.

In some countries a higher admission price can be charged if a 70mm print is exhibited. Because of this, many *mono* U.S. films can be heard and seen in 70mm only in such countries.

Of course, no such cheating is necessary with the *Star Wars* saga, and an honest-to-goodness six-track mix of every dubbed version of *Return of the Jedi* will be seen in at least a few 70mm prints, with Japan (subtitled) receiving over 15 prints. ■■■

THX SOUND SYSTEM — continued . . .

down a dB and a half," he says. "That's almost an octave better than most theater loudspeakers, which go down there but they always have a shelf. The horn works down to 80 [Hz] more or less, and then it takes a step down to where the reflex is working; you cannot afford the equalization to bring it back up because you run into [driver] excursion problems. And then, ultimately, it rolls off at 40 Hz."

Extended Low- and High-Frequency Response

Because of this extended low-frequency response, a switch was installed on the Neve 8108 console at Sprochet Systems, scene of post-production dubbing on *Jedi*, to simulate the "average bass" obtained in a theater, as indicated by a Dolby Labs' study of 150 first-run theaters. As Tom Holman notes, "Mixers must intelligently use the switch to know what problems will show up in a theater with a very flat low-end. There are two conflicting things that you want to know about: what it's going to sound like in the *real* world; and what you are actually putting on the track.

"There is a similar gain on the high-end, since the [JBL 2445 compression] driver used with the THX crossover follows the [ISO 'X'] curve, without extreme equalization, to 16 kHz. The typical high-frequency driver used standard multicell horns, follows the curve up to 8 kHz, and then it dies very rapidly.

"What is most noticeable between the THX system and a conventional system, when seated where the [latter] is at its best, is the extended treble response of the THX system in the octave above 8 kHz. The other thing is the lower distortion in the bass.

"For people seated in less desirable seats, the difference becomes more apparent. You can be sitting in a place where a notch caused by the lobes in the pattern of a multicell [HF horn] would create quite a large difference at 10 kHz."

The system is bi-amplified, and power amps used in a THX theater installation must meet the criteria that Holman set forth in his "New Factors in Power Amplifier Design" article in the July/August 1981 issue of the *Journal of the Audio Engineering Society*. (Before joining Lucasfilm, Holman had founded Apt-Holman, a company which makes the popular power amplifier/pre-amp combination.)

"We have 200 watts for the woofers, and 100 watts for the tweeters," Holman relates. "This dates back to what I learned when doing the powered Advent loudspeaker. People would say, 'Well, the woofer can take more power, so we put out this much power, and the tweeter can take this much less, so we give it this much.' You have this strange thing when

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you look at it on a spectrum analyzer, and see that during a cymbal crash the 10 kHz [third-octave band] is as loud as the 400 Hz. On a statistical, long-term basis that is not true; the high-end is rolled off. But in peaks like the cymbal crash, you need as much acoustic power available from the *tweeter* as you have on the low end.

"The reason why we have only 100 watts on the high end is that the tweeter is much more efficient. So we have, more or less, equal power handling across the band on the short term, at least to 10 kHz. You don't need it to 20 kHz."

Movie Theater Room Characteristics

After the installation of the crossover in the Lucasfilm dubbing stage during the Spring of 1982, the idea arose of marketing not only the crossover and the sound system, but also a complete theater inspection package that would be marketed under the tradename of the THX System. Four theaters have installed the THX system for presentation of *Return of the Jedi*: the Northpark and UA Prestonwood Creek 5 in Dallas, Texas; the Fashion Square in Orlando, Florida; and General Cinema's Avco in Los Angeles.

Prominent among the criteria that these and future THX theaters have to meet include specifications for reverberation time versus volume, noise level, screen illumination, and picture sharpness. "We look at a theater and make certain that the situation is not egregious [conspicuously bad]. We're not going to enforce the 80 line pairs per millimeter [picture] resolution that we get here in test situations, but we are trying to make sure that it's decent."

Since high-noise levels preclude the ability of patrons to hear low-level material on a track, Holman specifies that the equivalent NC [Noise Criteria] value of the theater be less than NC-30, and emphasizes that "this is the *required*, not the recommended value." (In one of the installations, Holman recalls, the NC level was reduced 15 dB merely by cleaning the air conditioning ducts!)

Any serious acoustics problems will be handled by an outside acoustical consultant that will reduce or increase, as the case may be, the reverberation time, and reduce the noise, to meet the required standards.

In the meantime, Lucasfilm staff will draw up plans to tailor the system to the individual room. "THX is more a concept of what a system should be," Holman emphasizes. "A specification of what it *should do* in the room, rather than a list of these drivers and those components. For example, if the theater has a balcony, then two high-frequency horns per channel might be used, with a switch installed to turn off the balcony horns when they are not needed."

One mandatory part of the THX system, in 70mm movie houses,

Close-up detail of THX sound system, a two-way configuration with a vented-box, direct-radiator LF design, and a compression driver/biradial horn for the top-end.
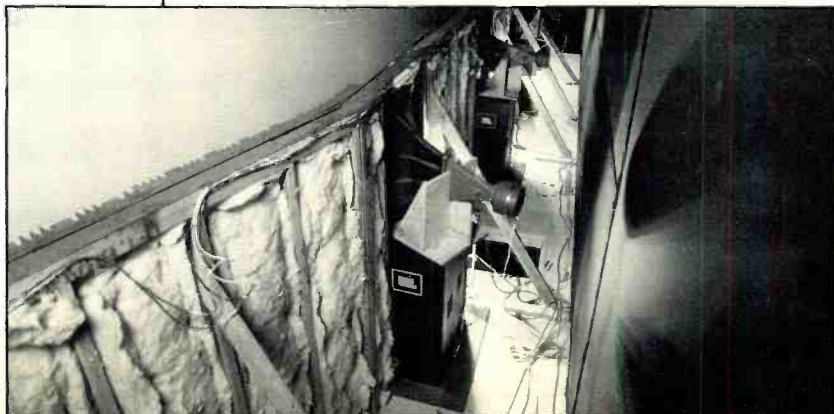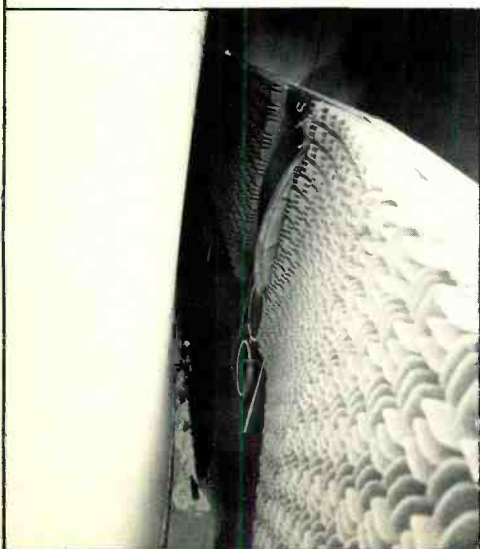
is the Kintek KT-9 subwoofer. Since speaker channels two and four are not normally installed in a 70mm THX installation, the subwoofer will be necessary to reproduce the low-frequency information on tracks two and four of Dolby 70mm prints; the subwoofer is optional in 35mm THX-equipped theaters.

Noting the most obvious way to reproduce low-bass information in a theater — "brute force, with a lot of available woofer cone area able moving a *lot* of air" — Holman chose the Kintek subwoofer for the elegance of its design. "It predicts the excursion of the driver, which is one limit, and the temperature of the voice coil of the driver, which is the other, and drives a limiter," he says. "Thus the subwoofer plays at very loud levels without audible distortion." Although it only uses two 15-inch drivers, Holman feels that one Kintek subwoofer is sufficient for theaters seating up to 1,000 patrons.

There are basically two choices that the theater owners have to make: one, in 70mm houses, to install amplifiers and speakers for channels two and four, which contain full-range information in only the handful of films released yearly in the "discrete" six-track format. [For further details, see "The Evolution and Utilization of 70mm Six-track Sound," by Larry Blake; *R-e/p*, April issue — *Ed.*] These speakers aren't needed in a THX installation for playback of standard 70mm Dolby "boom" prints, not only because of the presence of the Kinteck subwoofer, but also because of the increased bass-handling capabilities of the JBL speakers. Part of the Dolby 70mm format's design was to use speakers two and four to help compensate for the low-frequency deficiencies of standard horn-loaded theater speakers.

Similarly, theaters seating over 1,000 patrons might choose to have two bass cabinets per channel, for a total of four woofers. Holman notes that the standard two-woofers-per-channel will handle anything short of reproducing a live rock concert, which is hardly required to play back the great majority of stereo releases. Nevertheless, such an option is available, but "that extra 10 dB of headroom will cost you," he warns.

The recommended surround speaker for THX installations is the Boston Acoustics A-200, also currently in use on the dubbing stages at Sprocket Systems, Warner Hollywood, and Glen Glenn. "Boston Acoustics is interested in making a system with a tailored high-frequency response for motion picture theaters," Holman says, "which would have increased power handling, and follow the ISO 'X' curve in an average theater."

The cost of a THX system for a 35mm theater is expected to be in the area of $15,300, a figure that includes three loudspeakers (but not surrounds), THX crossovers (the only items made by Lucasfilm), construction of the speaker wall, amplifiers, and all consultation by Lucasfilm staff. Last, and most definitely not least, would be the license: the right to display the Lucasfilm THX logo in the theater lobby, and on the marquee. Every six months, from the date of the installation on, the theater will be inspected and must continue to meet the criteria, or risk losing its license.

Marketing of the THX system began in earnest this Fall.

Theater Alignment Program

When *Star Wars* appeared on the scene in 1977 with only about

THX SYSTEM — continued . . .

40 70mm prints, and played in 35mm Dolby stereo optical in 100 theaters, although all 35mm prints were in this latter format, there were only 300 Dolby Cinema Processors in the United States. (Just as when The Jazz Singer opened in 1927, there were only about 100 theaters equipped to handle sound.) Six years later the owners of the over 3,000 Dolby-equipped theaters were ready to sell their proverbial grandmothers for a print of Return of the Jedi, which would be released in over 150 70mm prints, the largest number ever made available in that gauge. Since the THX imprimatur only would be granted to four theaters, Holman wanted to make a concerted effort at helping the other 146 70mm houses present Jedi properly.

Sprocket Systems originated, and Twentieth Century-Fox approved, Theater Alignment Program (TAP), in which 29 technicians would be sent out in the field with 70mm rolls of Dolby alignment tone and pink noise, and inspect every house targeted for a 70mm print. In addition, an "800" Hot Line was established for any projectionist running Jedi to call if there were any questions or problems. This number was in effect until July 31, when, presumably, if things were not right by then, they never would be.

Before any of this could be done, however, the prints (and the film!) had to be made, and the standards that the technicians would enforce needed to be established. First order of business was to determine a reference level for all of the 70mm printing, which was to take place at three separate facilities. Before Jedi, the Dolby tone flux level was "uncontrolled," says Holman. "It was more or less 90 nWb/m, plus or minus 6 dB. That was because no one had ever really built the equipment to know what the flux level actually was; they would put on some striped film, turn it up until it distorted a certain amount, like 1%, and called that zero. This was reasonable considering the striping. However, they didn't know that the striping was pretty terrible, compared to what could be done."

What was done for Jedi was to have the 70mm prints striped at Film Processing Corporation, in Los Angeles. FPC has been making 35mm stripe materials since 1952, and early this year bought an old print-striping machine from MGM Laboratories.

Since 1974, almost all of the striping of 35mm and 70mm prints made in Los Angeles has been done by a single facility in Hollywood. Before that, most major studios had their own striping facilities, with the majority of them dating back to the days in the mid-Fifties when 35mm four-track mag prints were commonplace. FPC and this other company gave Holman 35mm four-track samples. Tests indicated that the FPC striping gave 5 dB more output.

Despite the initial good tests, the striping of 150 prints by a facility that had never striped one (though they had used an almost identical process in making 35mm single-stripe) is not, to put it mildly, a decision to be made without extensive testing. For example, a loop was played over 1,000 times on a dubber to check for oxide shedding. In the end, working in conjunction with engineers at Fox, Holman established a reference flux level of 185 nWb/m for Jedi 70mm prints.

Sound, as they say, is only half of the film, and similar detective work had to be done to secure copies of the SMPTE RP-91 70mm visual test film. Although the test chart had been approved in 1981, prints had not been struck from the original negative —not only because of the high cost of 70mm registration printing, but also (of course!) because of lack of interest in the part of the motion picture industry. It finally took the producer of Return of the Jedi, Howard Kazanjian, to set the bureaucratic wheels in motion to get the test film printed.

Sound System Alignment

Sprocket Systems eventually supplied each TAP technician with a 70mm SMPTE picture test film; 70mm Dolby Level test film; 70mm 10-kHz test film (to check head contact); and 70mm pink noise test film to set azimuth and to align the pre-amplifier response to within ±2 dB from 50 Hz to 12 kHz. In addition, 70mm degaussed film and a degaussing kit were supplied to facilitate checking the film path for magnetized components.

Each technician was expected to provide the standard tools of the trade: screen brightness meter; third-octave real-time audio spectrum analyzer; dual-trace oscilloscope; AC/DC voltmeter; and 35mm sound test films. The 35mm optical sound chain also was to be aligned in case, at any time during the run, the supplied 70mm print became damaged, and a 35mm print substituted.

Armed with this equipment, the TAP technicians proceeded to inspect the theaters that had requested 70mm prints, with only about 10 theaters finally being unable to meet Lucasfilm's picture and sound criteria.

Apparently, there is a theater in Alaska ("It's probably called the 'Polar I,' " says Burtt, envisioning a drawing of steam on the marquee saying: 'It's warm inside!') currently showing Return of the Jedi in 70mm, even though it only has one projector, and no platter system. Supposedly, the audience waits patiently between reels for the changeover. Holman notes that "The only power we had this time around was to not give the theater a 70mm print. Hopefully the next time around we will have contractual power to say that you have to have the correct number of footlamberts on the screen, and things like that. But it is much trickier when there are technical matters in contracts."

After a magnetic print is striped, the film "ideally but not necessarily," says Holman, sits around for a few days while the oxide settles ("cures"). Then each print is sounded in real-time from a six-track printing master. As is standard operating procedure in Hollywood, every 70mm print of Jedi was listened to in a theater. On both The Empire Strikes Back and Return of the Jedi, one in five prints was rejected for some picture or sound fault. Common sound errors include sync, dropouts and, with picture, various and sundry blotches.

Usually someone closely associated with the film is present during print checking. On Raiders of the Lost Ark, for example, supervising sound editor Richard Anderson checked 50 prints; producer Howard Kazanjian personally inspected 65, 70mm prints of Jedi.

If something is found to be wrong with a 70mm print, a decision has to be made as to whether it should be placed on a low priority, and sent to a theater in the boondocks; whether the reel has to be re-sounded; or, if the problem is very serious, rejected en toto. At the cost of $14,000 per 70mm print for Jedi, the scrapping of a reel is not a decision to be taken lightly. If the only problem on a reel is a flash on one frame, should that reel be sent to Spearfish, North Dakota, or should it be rejected?

In the race to get such a large number of 70mm prints out, a few theaters received prints with no sound. It is debatable whether this is better, or worse, than sounding reel #6 with the printing master of reel #5 or, as happened during the sounding of the initial Star Wars 70mm prints, using the soundtrack of Fantasia. (This problem was caught when the print was being checked, it should be noted.)

Tom Holman is quick to point out that inspecting theaters is only just part of the battle: Return of the Jedi was made to real-world specs. For example, even though SMPTE recommends that theaters have a light level of 16 footlamberts on the center of the screen, there probably isn't a theater outside of a laboratory screening room that is this bright. With this in mind, Jedi was timed and color-balanced for 12 footlamberts.

In regard to the sound, a test mix was played in five theaters to determine the average level of the boom channels, which varied widely depending on whether a theater had subwoofers or not, and what kind of speakers and amps were used for channels #2 and #4. (What was learned from these tests regarding the dynamic range of the mix has been outlined previously.) In the TAP instruction manual, sprocket systems detail the levels to which the various combinations of speakers should be aligned to properly replay the boom information.

Holman is generally pleased with the results of the TAP program, although he still has the sneaking suspicion that some theater, somewhere, is playing Jedi with the center and surround channels reversed, all to a blissfully contented audience. □□□
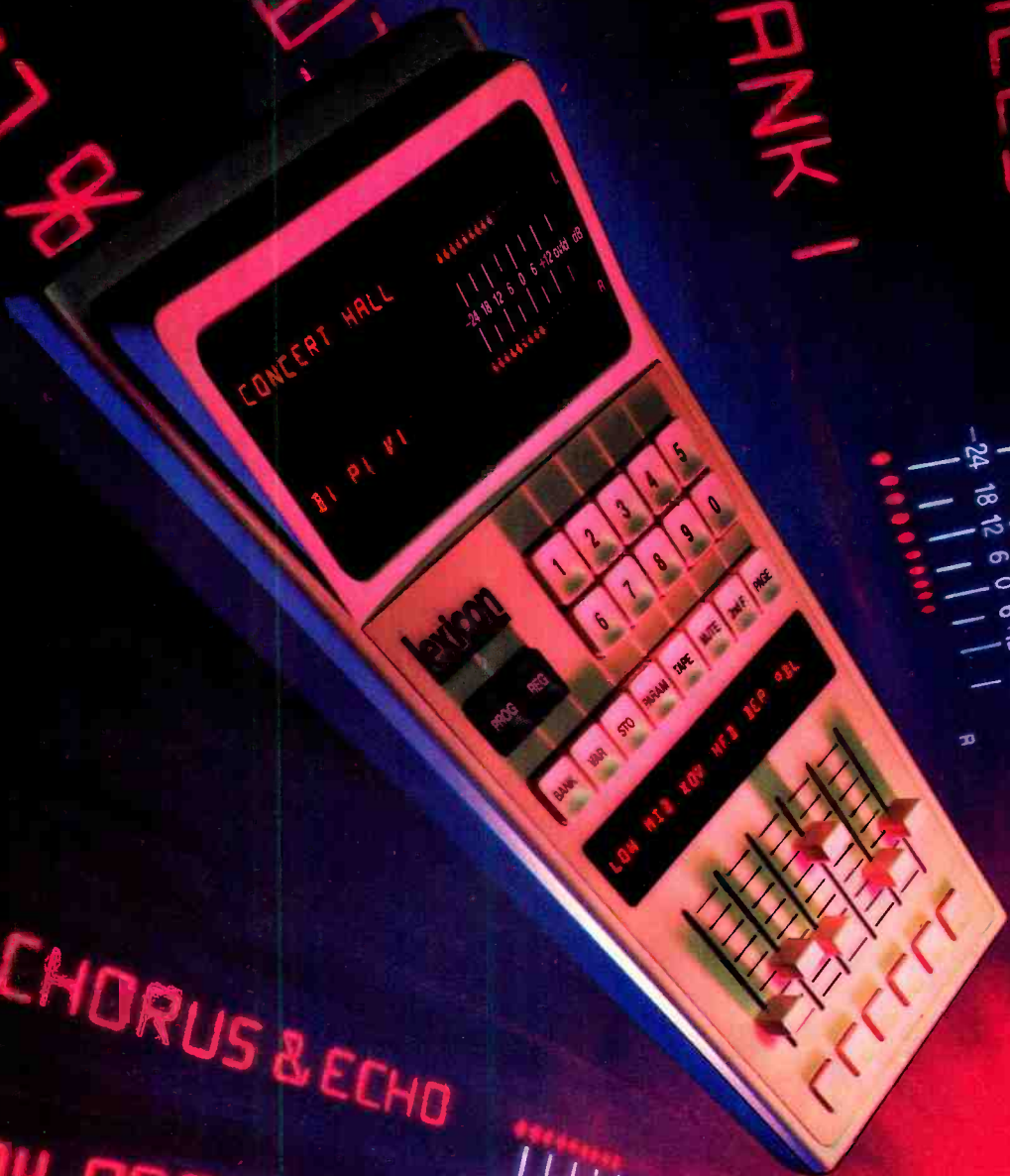
North East:

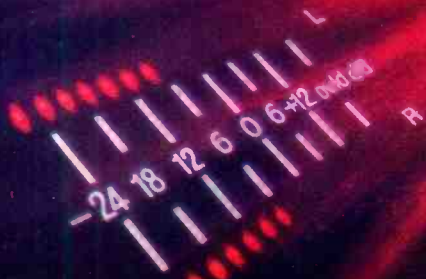☐ **ROAR PRODUCTIONS** (Columbia, Maryland) has installed a new Neotek Series II console featuring four-band state variable equalization, and light column metering on all channels. The installation was completed by Roar's technical staff in mid-October, and included complete rewiring of the control room and relocation of outboard equipment to afford a more comfortable and efficient recording environment. A DeltaLab digital delay unit, Klark-Teknik third-octave equalizer, and fourth 2-track recorder were also recently added. *6655H Dobbin Road, Columbia, MD 21045. (301) 596-0600.*

☐ **UNIQUE RECORDING** (New York City) has updated its in-house instrument selection with the recent addition of a Yamaha DX-7 digital synthesizer, PPG Wave 2.2 digital synthesizer and sequencer, and a second E-mu Systems Emulator equipped with J.L. Cooper Emulator-to-MIDI modification. Equipment acquisitions include a Sony DRE 2000 digital reverb, and a set of eight Neve 1077 mike pre-amp and equalization modules. *701 Seventh Avenue, New York, NY 10036. (212) 921-1711.*

☐ **WESTRAX RECORDING STUDIOS** (New York City) recently upgraded its 8-track facility to 16-track with the installation of a Sound Workshop Series 30 console and Tascam 85-16B 16-track tape machine with dbx noise reduction. Other recent additions include a Lexicon PCM-42 delay line, Neumann U-87 microphone, an Oberheim OBXa synthesizer and a Linn Drum Machine. Two- and 8-track recording services continue to be available. *Manhattan Plaza, Basement Level, 484 West 43rd Street, New York, NY 10036. (212) 947-0533.*

☐ **INNER EAR RECORDING** (Queens, New York) has just installed a Studio Technologies Ecoplate II reverberation plate in its Otari/Tangent-based 16-track recording facility. Calibration Instruments MDM-4 and Tannoy SRM12B "close-field" monitors have been added to UREI 811A and Auratones. Also recently acquired is a vintage Steinway 6'6" grand piano to complement a large list of in-house instruments. *118-17 97th Avenue, Queens, NY 11419. (212) 849-5725.*

INNER EAR — New Monitor System

☐ **GREENE STREET RECORDING** (New York City) has updated its outboard equipment to include a Lexicon 224X digital reverberation unit, a pair of UREI LA3A limiters, a pair of Lexicon PCM-42 delay lines, an Eventide H949 Harmonizer, and an Aphex Aural Exciter. The digital instrument collection now includes an Emulator, a DX drum machine, and an E-mu Systems Drumulator. Renovation of the lounge area has been completed to, according to owner **Steve Loeb**, create a more expansive space characteristic of the studio's Soho neighborhood. *112 Greene Street, New York, NY 10012. (212) 226-4278.*

☐ **NIMBUS NINE RECORDING** (New York City), formerly a private studio, has opened its doors to outside clients. Owned by **Geoff Daking**, the studio features a Trident Series 80 console, MCI JH-24 multitrack, JBL 4430 monitors, EMT tube stereo reverb, AKG two-channel reverb, Lexicon PCM-42s, Pultec equalizers, API and ADR compressor/limiters, Ursa Major Space Station, Valley People Kepex IIs and Gain Brain IIs, plus SMPTE interlock. Room equipment includes a Yamaha grand piano, Fender Rhodes and Ludwig Drums. There is also a Linn Drum Machine and a Sequential Circuits Prophet 5 available on a rental basis. *1995 Broadway, New York, NY 10023. (212) 496-7771.*

☐ **TIKI RECORDING STUDIOS** (Long Island, New York) has completed installation of a new Trident TSM console in its newly opened studio complex. The console features 40 inputs, a 32-track monitor section, and has many modifications, according to owner, **Fred Guarino**. *186 Glen Cove Avenue, Glen Cove, NY 11542. (516) 671-4555.*

☐ **CLINTON RECORDING** (New York City) has installed two new custom-built Neve Model 8078 consoles, according to co-owners **Bruce Murley** and **Edward Rakowicz**. Both consoles have 40 inputs and 24 output busses, and one is equipped with NECAM II automation. According to Murley, "These boards will often be used for jingle recording and mixing. An engineer working in jingles has to be both good and fast, and the layout of these Neves is extremely conducive to efficient jingle recording." Both custom consoles are equipped with a separate monitor section, larger equalizer knobs, and several other features. Multitracks are two Studer A800s, plus four A80 and four A810 mastering machines in both control rooms. The downstairs studio has a floor area of 1,000 square feet, and the larger room upstairs fills 2,000 square feet with a 22-foot ceiling. Acoustic design was by Rakowicz and architect **Maurice Wasserman**. Monitors are UREI 813s powered by McIntosh amplifiers. Outboard gear includes several EMT 140 reverbs. *653 10th Avenue, New York, NY 10019.*

Southeast:

☐ **MUSCADINE STUDIOS** (Macon, Georgia) recently added an Ampex MM-1200 16-track recorder to its Tascam, Studer, and Revox tape machines. The updated 16-track facility centers around a Flickinger console, and JBL 4312 studio monitors. Outboard equipment includes Lexicon Prime Time, Lexicon PCM-41 delay line, MXR digital delay, dbx and Orban compressor/limiters, and an AKG BX-20 reverberation unit. Studio owner/manager is **Paul Hornsby**. *3078 Vineville Avenue, Macon, GA 31204. (912) 745-2401.*

☐ **MORRISOND RECORDING** (Tampa, Florida) has opened Studio B. The new room is equipped with a Soundcraft Series 400B console and an Otari MX5050 8-track tape machine, and was designed for pre-production and voice overs. The 24-track Otari/Sound Workshop equipped room is also still in use. *5120 N. Florida Avenue, Tampa, FL 33603.*

☐ **CRITERIA RECORDING STUDIOS** (Miami, Florida) has recently added an Ampex ATR-100 2-track machine with ¼- and ½-inch heads available, an Otari MTR-90 24-track machine with autolocator, and has acquired its second Mitsubishi X-80 PCM 2-track digital recorder. The two digital systems are available in any of the four studios within the audio/video and disk-cutting complex. *1755 NE 149 Street, Miami, FL 33181. (305) 947-5611.*

☐ **NEW RIVER STUDIOS** (Fort Lauderdale, Florida) has recently upgraded the main monitor system by installing a pair of UREI 813B monitors. The control room is equipped with a Neve 8108 48-track console with NECAM II automation, a pair of Studer A800 24-track recorders, and an Audio Kinetics Q-Lock synchronizer for audio post-production. *408 South Andrews Avenue, Fort Lauderdale, FL 33301. (305) 524-4000.*

CRITERIA — Second Mitsubishi X-80

☐ **STARTEC** (Washington, D.C.) has opened a new 24-track facility featuring a custom built API 32×24 console purchased from Sunset Sound in Hollywood, CA. Installation and control room design was done by Wolff Associates and Borge Systems Ltd. Tape machines are Stephens 24- and 4-track ½-inch, Otari MTR-12 ½-inch stereo, and Ampex 440. Outboard equipment include delay units by DeltaLab, Lexicon, Ibanez, MXR, and Yamaha; compressor/limiters from dbx, LT Sound, B&B Audio, UREI, and API; and Lexicon 224 reverb. Synclavier II, Prophet 5, and MemoryMoog synthesizers along with Yamaha grand and Fender electric pianos comprise the in-house keyboard instrument list. Owners are **Nick Koumoutseas** and **Jim Wallace**. Staff is **David Hanbury** chief engineer; **Kevin Hayes**, arranger/producer; and **Paul Wolff**, engineer/systems engineer. *1737 DeSales Street SW, Washington D.C.*

South Central:

☐ **BRAZRO RECORDINGS** (Chattanooga, Tennessee) is a 16-track facility that has recently added equipment for real time cassette duplication. The studio has also prepared its own "ear training course in equalization," which it offers for sale in cassette format. *1215 N. Concord Road, Chattanooga, TN 37421.*

South Central (continued):

☐ **THE CASTLE RECORDING STUDIO** (Franklin, Tennessee) has installed a 3M 32-track digital mastering system to complement the facility's Harrison 48-channel 3232C console, and two Studer A80VU analog tape machines. *Old Hillsboro Road, Route 7, Franklin, TN 37064. (615) 791-0810.*

☐ **JIMMY SWAGGART MINISTRIES** (Baton Rouge, Louisiana) has opened a new broadcast complex equipped with five Neve consoles. According to **Dave Cooper**, director of television operations and engineering at the new facility, a Model 8128, 32 input/24 bus console equipped with NECAM II automation has been installed in audio control room A, while control room B houses a 24-bus Model 5116 fitted with 36 inputs. These remix rooms are also equipped for overdubbing and live recording. Each of two identical video edit suites is equipped with Model 5412 12/4 consoles. The VCAs fitted to these two consoles are tied into the computer editing system, enabling audio/video synchronization. The third 54 Series 16/4 console is installed in another control room for studio video productions. Acoustic design of control rooms and studios was performed by **Jerry Milam** of Milam Audio. Audio control rooms are equipped with UREI 813 speakers powered by UREI amps. Multitracks comprise two Studer A800 24-tracks, and four A810 stereo machines. *8919 World Ministry Avenue, Baton Rouge, LA 70810. (504) 769-8300.*

Mid-West:

☐ **BREEZEWAY STUDIOS** (Waukesha, Wisconsin) has updated its equipment list with the additions of an MCI JH-110A ½-track mastering deck with Dolby A noise reduction, two UREI 1176LN peak limiters, an Eventide H910 Harmonizer, a set of Omnicraft GT-4 noise gates, five pairs of AKG K240 headphones, and a pair of White 4100 third-octave equalizers for room equalization. UREI 813B studio monitors are being installed in the main control room. The studio is also adding a new lounge area highlighted by a viewing window into the recording studio. **Paul Wehrley**, former bass player for Arroyo and Rio, has signed on as studio manager and engineer. *363 West Main Street, Waukesha, WI 53186. (414) 547-5757.*

☐ **SOUNDSMITH RECORDERS** (Indianapolis, Indiana) has added **Jeff Bowen** to its staff as production co-ordinator and marketing director. **Al Thompson** has joined the sales staff, and will serve as an audio consultant and sound engineer for upcoming live performances. *5210 East 65th Street, Indianapolis, IN 46220. (317) 842-4905.*

☐ **SONIC ART** (Lake Villa, Illinois) is a new 16-track facility featuring MCI, Soundcraft, Lexicon, Crown, and UREI equipment. It also has three isolation chambers. *23783 W. Petite Lake Road, Lake Villa, IL 60046. (312) 356-8992.*

Mountain:

☐ **APPLEWOOD STUDIOS** (Golden, Colorado) has acquired a new Studer A800 MK III 24-track tape machine to complement the existing Neve 8036 console. Mark III Electronics and ½-inch capability have been added to the Studer A80 mastering machines, and the facility has added 10 API 550 equalizers, and six API 500 compressors. The copy room has been updated with the addition of four new Studer A710 real time cassette decks, and two Revox PR99 copy machines. The Colorado Audio Institute teaches classes at the studio when its not in use for record dates. *680 Indiana Street, Golden, CO 80401. (303) 279-2500.*

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Southern California:

☐ **THE COMPLEX** (Los Angeles) has installed GML Research's newly-developed moving fader automation system in Studio C. The system, developed by George Massenburg, breaks fader movement into more than 1,000 segments, and takes less than 80 milliseconds to move the fader from one end of its travel to the other. The new system has a sampling rate of four times a frame using SMPTE timecode. The fader automation has no VCAs and works out of random access memory rather than floppy discs for an access speed that is said to be 30 times faster than other systems. The automation package is designed to follow an engineer's movements during a mix, and take direction from the multitrack instructions and SMPTE locations. *2323 Corinth Street, West Los Angeles, CA 90064. (213) 477-1938.*

☐ **FRED JONES RECORDING SERVICES** (Hollywood) has upgraded its 24-track facility to become a complete audio post-production facility with the addition of a new BTX Softouch™ System. The Softouch system consists of three intelligent, distributed modules: the Softouch editor/controller, Shadow II™ synchronizer and a modular timecode system known as Cypher™. These modules are networked together via RS-232, while transports and other studio equipment are connected via straightforward interfaces. The facility now offers complete multitrack sweetening for video projects. The new system is said to be the first BTX Softouch system on the west coast; installation was handled by **Ana Wilczynski** of Everything Audio, and **Jim Lucas** of BTX. *6515 Sunset Boulevard, Suite 205, Hollywood, CA 90028. (213) 467-4122.*

☐ **SYNAPSE RECORDING** (Burbank) is the new audio recording facility of Nicholas Simone Productions. The studio features an Otari 8-track recorder, and Synclavier II and Yamaha DX7 synthesizers. For scoring, 16 digital tracks are provided by use of the Synclavier II in conjunction with the Otari 8-track. The studio is available for jingle production, sound recording and synthesis. *444 S. Victory Boulevard, Burbank, CA 91502. (213) 661-7777.*

☐ **PREFERRED SOUND** (Woodland Hills) is a new 24-track recording facility with living accommodations available. The control room centers around an AMEK 2000 Series II transformerless console with full parametric equalization, linked to an Otari MTR90 24-track recorder with autolocator. The monitor system is comprised of UREI Time Align, JBL 4401, and Auratone monitor speakers. Outboard equipment includes Lexicon Prime Time, Eventide Harmonizer and Flanger, Lexicon 200 Digital reverb, MXR and Roland Stereo Flangers, Valley People Kepex Noise Gates, dbx 165 and UREI LA-4 limiters, and UREI 1176 limiters. Among the instruments available, the studio boasts a fully restored 1860 Bosendorfer grand piano. Owner is **Scott Borden**, former partner in Norman's Rare Guitars. *22700 Margarita Drive, Woodland Hills, CA 91364. (213) 883-9733.*

☐ **AUDIO CYBERNETICS** (North Hollywood) has just added an E-mu Systems Drumulator to its music composing and production facility. Composer/owner **Christopher Currell** states, "Our facility — based on the Synclavier II Digital Synthesizer, as well as other synthesizers — gives the composer or producer an almost limitless capability for scoring and producing music and sound effects." Currently on order is a second, larger Synclavier II, a Lexicon 224X digital reverb, a Linn Drum Machine, a Sony PCM-F1 digital processor for sound-effects recording, and a Digital Guitar interface for the Synclavier. *5102 Vineland, North Hollywood, CA 91505. (213) 760-8333.*

☐ **SUNSET SOUND** (Hollywood) has placed a new custom-built consolette into service. The consolette, a 12-channel version of the studio's upcoming Studio 1 custom-built console, features digital logic controlled foldback, echo, program and stereo bus assigns, and silent inserts and muting. Jensen transformers and Jensen 990 op-amps were used in all gain stages. New general manager, **Craig Hubler**, says "We continue to build our own consoles as we feel there are none available that fulfill both our clients demands for more flexible equipment and Sunset's longstanding reputation for having the cleanest and quietest boards possible." Welcome back to traffic manager **Gail McCabe** who has returned to the studio after an illness. *6357 Selma Avenue, Hollywood, CA 90028. (213) 467-2500.*

☐ **LARRABEE SOUND** (Los Angeles) has expanded Studio A's Solid State Logic Series E console to provide 56 channels for use with Ampex MM-1200 and Studer A80 24-track tape machines. Studio A is also equipped with an Audio Kinetics Q-Lock 310 synchronizer, and features a monitor system by George Augspurger. *8811 Santa Monica Boulevard, Los Angeles, CA 90069. (213) 657-6750.*

☐ **A&M RECORDING** (Hollywood) has purchased three Studer A-800 MK III 24-track tape machines for Studios A, B and D. In September, A&M hosted a Digital Synthesizer Forum organized and presented by the Los Angeles Chapter of NARAS. *1416 North La Brea Avenue, Hollywood, CA 90028. (213) 469-2411.*

☐ **BELL SOUND** (Hollywood) has taken delivery of two Studer A810 tape recorders. The machines are two-track stereo recorders with a center timecode track, and were provided by Audio Engineering Associates. Bell Sound is a 24-track facility specializing in television

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— SOUTHERN CALIFORNIA FACILITIES SPOTLIGHT —

PREFERRED SOUND

AUDIO CYBERNETICS

SUNSET SOUND — New Custom Consolette

LARRABEE SOUND — SSL Series E Console

FUTURE DISC — Mitsubishi X-80A Digital

commercials and jingles production. *916 North Citrus Avenue, Hollywood, CA.*

☐ **FUTURE DISC SYSTEMS** (Hollywood) has installed a new Mitsubishi X-80A digital two-track, and companion DDL-1 digital disk-cutting preview unit. *3475 Cahuenga Boulevard West, Hollywood, CA 90068. (213) 876-8733.*

Northern California:

☐ **T&B AUDIO LABS** (San Francisco) has added MDM TA2A Time-Aligned™ Nearfield™ monitors, and Ellinger quad noise gates that can be synced to one another, or to an external trigger source, in any combination. Individual control voltages are brought out to patch points, so that other voltage-controlled devices can follow the envelope of the gates, making them ideal for synthesizer work. *3018 22nd Street, San Francisco, CA 94110. (415) 821-3065.*

☐ **RUSSIAN HILL RECORDING** (San Francisco) in partnership with **Persistent Image**, has opened a new film to tape transfer studio. This new facility, Russian Hill Film To Tape, generates high quality transfers from 35mm or 16mm film positives to videotape format. Developed with the assistance of **Clark Higgins**, designer of video transfer systems for both Lucasfilm and Zoetrope studios, the new system is also capable of generating frame accurate timecode in a multitude of formats: vertical interval, longitudinal, window dub, film frame numbering, in addition to SMPTE, film-edge codes, reel #, scene and take information, etc. Since the studio's KEM K-800 is a six-plate flatbed editing machine, Russian Hill can offer scratch mixes from two full-coat magnetic tracks, or from four 35mm magnetic tracks. Release prints with optical sound and news style film (i.e. single-system magnetic stripe films) can also be transferred. Russian Hill Film To Tape was used extensively during post-production work for *The Right Stuff.* Says supervising editor Glen Farr, "The KEM system saved us a lot of time. We could mix music, dialog, and effects simultaneously, while the transfer was being made. It is a very flexible system." *1520 Pacific Avenue, San Francisco, CA 94109. (415) 474-4520.*

RUSSIAN HILL FILM TO TAPE

☐ **PATCHBAY STUDIOS** (San Rafael) has installed full level and mute automation in its 24-track Studio A. The 30-channel VCA system also provides automated master fade and effects. A custom interface also enables automated mixes in the 8-track Studio B. New in the computer corner is the Decillionix DX-1 digital sound effects program for the facility's AlphaSyntauri poly synth/composer. *2111 Francisco Boulevard, San Rafael, CA 94901. (415) 459-2331.*

☐ **FANTASY STUDIOS** (Berkeley) has installed its third Neve 81 Series console in a totally remodelled and rebuilt Studio C. According to executive vice-president **Roy Segal**. "Client acceptance of our two 8108s has been overwhelming. Neve has contributed to improving our sound, and we wanted another one just like the other ones." In contrast to the 8108s in Studios A and B, which are equipped with NECAM automation, the new 32×24 console features VCA grouping. Acoustic and electrical design of Studio C were performed by Fantasy's own engineering staff. *10th & Parker, Berkeley, CA 94710. (415) 549-2500.*

Bahamas:

☐ **COMPASS POINT** (Nassau) has re-equipped Studio B with a 40-channel Solid State Logic desk equipped with Total Recall automation, a new Studer A800 24-track, and UREI 813B monitors. Studio A is also being refurbished with the installation of a 36-channel MCI JH-536 console. *P.O. Box N4599, Nassau, Bahamas. (809) 327-8282.*

Canada:

☐ **NATIONAL FILM BOARD OF CANADA** (Montreal, Quebec) has installed a custom Neve Model 51 film mixing console. "The entire configuration of this desk was specified by our engineers," says **Leonard A. Green**, NFB chief of operations. "Even before we brought in our electrical engineers, we spent two or three months talking with our staff mixers about what they needed in a mixing board." In particular, NFB requests included an atypical electical layout: that the compressor be placed *after* the input amp and fader. "This desk has some 30-odd compressors, one in each input module," says **Peter Strobl**, one of the studio's mix engineers. "During a film mix [as opposed to a music mix], an input may see dialog or sound effects from several different sources, each one different in level or equalization. Unlike in a music studio, we do not have the time to optimize compression for each source. But, since the compressor follows the EQ and fader, we can expect the threshold setting to remain fairly constant throughout a mix." The Model 51 is equipped with eight plasma peak-reading meters: six for the main out, and two for combined purposes. "We find that the peak-reading meter is far superior when transferring to optical sound," Strobl offers. "Because the PPM's characteristics are very similar to the galvanometer on the optical recorder, we can make 0 dB on the PPM correspond with 100% optical modulation." The desk is also automation-ready, with VCAs on each input module, and two VCA submasters. "The VCAs help, especially in stereo mixing," Strobl continues. "An assistant engineer can be premixing a subgroup of stereo effects, which I can then manipulate with a single VCA master. In effect, the main engineer has an override on what the assistant is mixing." *Montreal, Quebec.*

Great Britain:

☐ **THE CHURCH** (Crouch End, London) The **Eurythmics'** personal-use studio has intalled a Soundcraft Series 2400 28/24 console, to replace the band's original Soundcraft Series 2 16/8. The group are using the new 2400 with a separately mounted channel taken from the old Series 2, saved for its "unique tone." The desk is linked to a Soundcraft SCM 24-track. *Crouch End, London.*

☐ **THE BRITISH BROADCASTING CORPORATION** (London) has installed its eleventh Solid State Logic Master Studio System, which will provide live transmission and simultaneous multitrack recording from the BBC's Hippodrome Theatre is Golders Green, London. The custom L-shaped mainframe houses 40 input/output modules in its main section. The wing section houses 16 more modules, plus an additional 30 SSL microphone pre-amp modules. The arrangement provides a total of 120 line inputs, 86 mike inputs, 56 four-band parametric equalizers, 56 compressors plus a main program compressor, 56 SSL expander/noise gates and eight VCA Group Masters. SSL has also supplied a remote producer's console and patchbay with the system. The system is fitted with the SSL Primary Studio Computer and Real Time System, which provide computer assistance both during live transmission and subsequent multitrack sweetening. *Broadcasting House, London W1A 1AA.*

Puerto Rico:

☐ **DOUBLE TALK** (Puerto Nuevo) a wholly-owned subsidiary of Crescendo Audio Productions, is a new state-of-the-art dubbing stage equipped with a Spectra Sonics 1024 console, MCI JH-24 24-track, JH110B-2, JH110B-4, and Studer A810 mastering decks, UREI 811 and Auratone monitors, and a variety of Magna-Tech pick-up recorders and 16/35mm dubbers. Outboard equipment includes Dolby racks, an Audio Kinetics Q.Lock 310 synchronizer, Eventide H949 Harmonizer, Delta Lab DL2 Acousticomputer, Vally People Gain Brain II, Kepex II, and Dynamite 410-2 limiter/gate, dbx 160 and 165 compressor/limiters, Ashley SC68 and Audio Arts parametric notch filters, and a Pultec effects filter. *Calle Constitucion, Num 707, Puerto Nuevo, PR 00920. (809) 792-6466.*

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1/2" GLASS
GRILLE CLOTH
HARDWOOD TRIM

3/8" GLASS
1/4" GLASS
1/2" GLASS
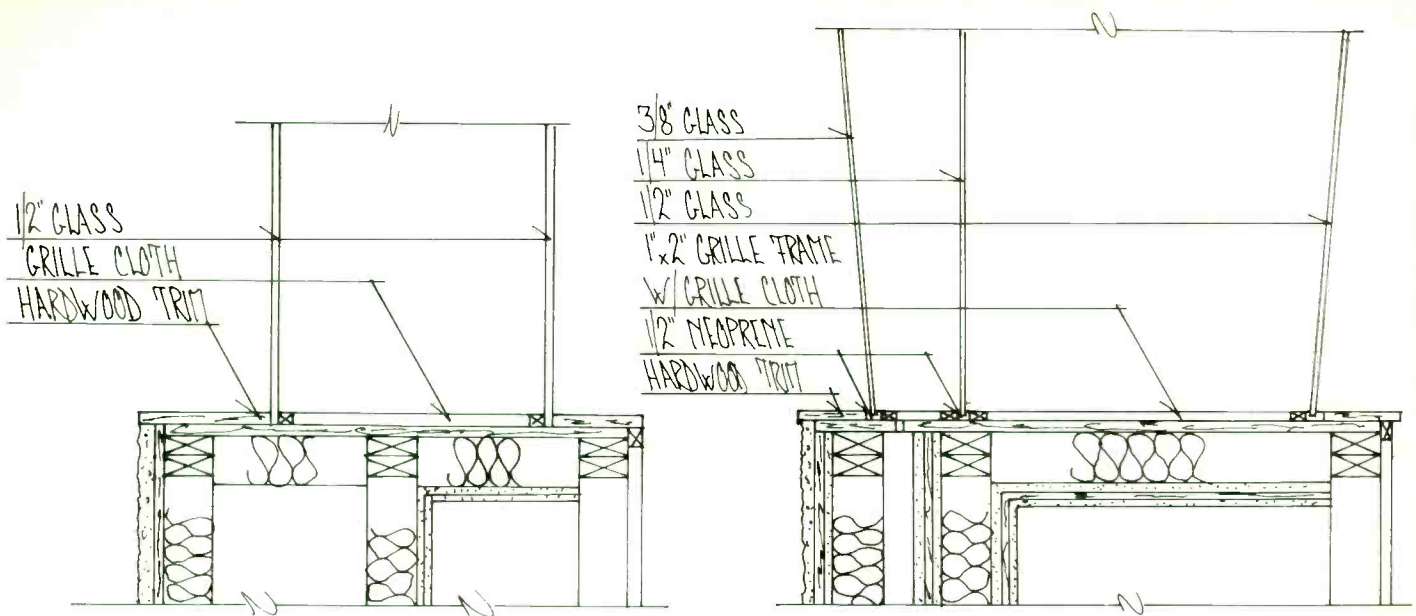1"x2" GRILLE FRAME
W/ GRILLE CLOTH
1/2" NEOPRENE
HARDWOOD TRIM

Figure 1 (left): control room/studio window before reconstruction; and Figure 2 (right) the final triple-glass design to ensure adequate sound isolation.

complete. Cosmetically, Rocshire's facility was on a level equal to the best anywhere; unfortunately, that was where it ended. The original contractor, doubling as supposed acoustical expert, made just about every possible mistake in the areas affecting isolation performance. And such problems are the most difficult to correct after the fact, both from a cost as well as an emotional standpoint. Would you want sawdust on your Steinway?

Correcting the Deficiencies

At Rocshire, two major isolation-related problems existed: firstly, insufficient sound isolation from the common control room and studio wall; and secondly, noise and vibration transmission from an adjacent wood-working shop that had been sited on a floor slab and wall common to the studio. Correcting the vibration problem came down to a simple choice: either tear down the existing studio and start from scratch with a floated slab; or buy out the neighbor's lease, and live with the problem in the interim. Since additional space was needed for a planned expansion anyway, there was no question as to the solution here — get rid of the neighbor . . . and his noise. Once he was gone, it would be a simple matter to improve the wall construction from that side. Obviously, having control over adjacent space is the surest way of avoiding the present or future use of extreme sound isolation measures.

Insufficient isolation between Rocshire's control room and studio was another matter, however. The existing wall and window construction was totally inadequate; in addition, a change in control-room monitoring had been requested which in any case required reconstruction. Here the only sensible solution *was* to tear down the structure and start over.

Two panes of glass originally had been used to form the conrol-room window. In some situations, this arrangement may be adequate, but not if there is total coupling between them, as illustrated in Figure 1. During reconstruction, three separate sections of glass of different thickness were used, consisting of nine different pieces supported by four separate walls, each with multiple layers of sheetrock and soundboard, and lots of silicone sealant (Figure 2).

With the vibration and control room to studio isolation problems resolved, approval was given by Rocshire staff to attack other weak acoustic areas. While demolishing the wall between the control room and studio, a good picture of how the facility had been constructed previously became available for the first time. (There were no plans available for what was originally built.) This newly revealed information clearly indicated several areas where a substantial
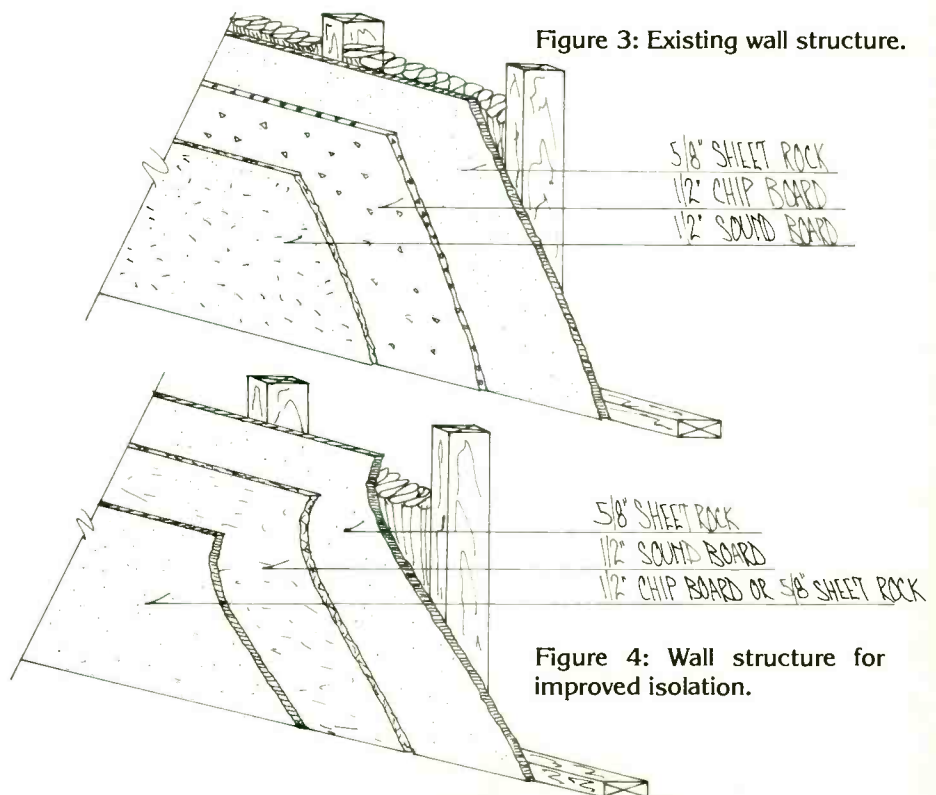
Figure 3: Existing wall structure.

5/8" SHEET ROCK
1/2" CHIP BOARD
1/2" SOUND BOARD

5/8" SHEET ROCK
1/2" SOUND BOARD
1/2" CHIP BOARD OR 5/8" SHEET ROCK

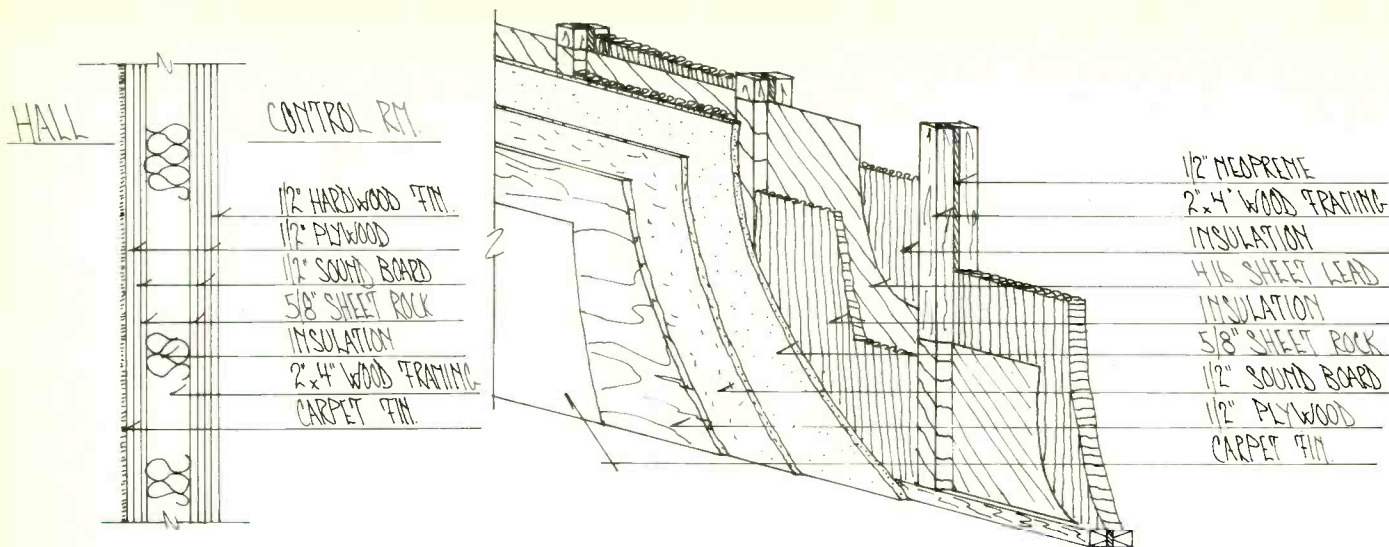Figure 4: Wall structure for improved isolation.

Figure 5 (left): section of hall/control room wall after reconstruction; and Figure 6 (right) a sectional view showing 4-pound lead sheet wrapped around and between wood framing.

improvement in acoustic treatment could be made.

In the existing construction, the basic wall system, shown in Figure 3, comprised of sheetrock, followed by a layer of chipboard, plus a final layer of soundboard. Far better results could have been obtained by reversing the order of materials, as shown in Figure 4, with the soundboard layer sandwiched between the sheetrock and chipboard. Wherever the former situation was encountered, an extra layer of sheetrock was added to increase the transmission loss of that particular wall and to take advantage of the soundboard layer already in place.

Which illustrates the fact that there is more to obtaining good sound isolation than simply relying on mass. Both walls (without the extra layer of added

sheetrock) have the *same* mass, but exhibit entirely different isolation performance. In addition to mass, bending stiffness and internal damping are other acoustic factors that come into play. And which is why an insulated wall with multiple layers of sheetrock can do in five or six inches what it takes a solid concrete wall 12 to 24 inches to achieve, in terms of sound absorption. Part two of the Mass Law is the Law of Diminishing Returns; after the first doubling of the mass, which increases the transmission loss by as much as 5 dB, each additional mass doubling produces, in practice, progressively less improvement per unit weight, and at increasing cost.

The other problem with adding mass to provide adequate sound isolation is that it always consumes space. The

Figure 7: Section of studio door design.
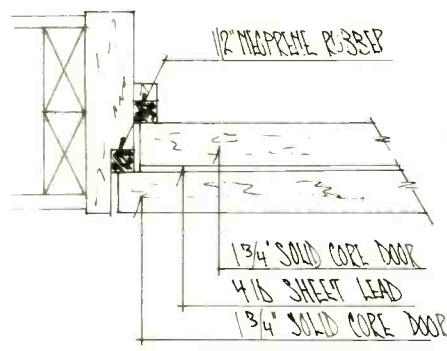
complex double (and sometimes triple) wall construction typically utilized in studio construction is the only way of achieving the degree of isolation required within a reasonable amount of real estate. In new construction, this forms one of the primary design parameters. In reconstruction, however, there are times when even an extra inch of spare space simply does not exist.

Figures 5 and 6 show how a substantial improvement was made, despite the fact that there just was no additional space to add more mass. The intention here was to increase the transmission loss of sound across the wall separating the control room and studio entrance hall. In this instance, the control room side could not be changed without serious cost penalties. Instead, all work had to be carried out on the hall side, which was stripped down to the 2×4 studs. After verifying that this wall was non-loadbearing, the existing studs were cut lengthwise in two, while still in place. After removing an additional half-inch of lumber, and replacing it with neoprene, the new 2×2 studs were put back into the wall. Now the only sound transmission path occurred at the top and bottom of the wall.

Ideally, additional layers of sheetrock

The narrow lobby and hallway prevented external sound treatment of studio walls. Layers of neoprene and lead sheet reduced sound transmissions.

The recording area feature a 1910 Steinway grand.

determines how the room will sound, was addressed. Although such a scheme would reduce still further studio floor space, it was decided to splay the walls slightly to minimize potential flutter-echo problems, as shown in the accompanying floor plan. Again, due to space limitations, all sound trapping had to be done in the ceiling.

In many studios, specific areas are set aside to be either absorbent or reflective, to provide different recording environments for enhancing the sound of specific instruments. In a small room, this type of flexibility can only be provided by surfaces designed to be acoustically variable. In any size room, the easiest surface to make variable is the floor. Simply install a hardwood floor, and provide carpets when a less reflective condition is required. There are several ways of varying wall surfaces, ranging from simple manual operation, to remotely controlled, motor-driven devices. Figure 8 illustrates an inexpensive, long-used technique that was employed at Rocshire to enable some of the wall surfaces to be varied from hardwood panelling to fabric-covered insulation (Owens-Corning #703).

In the control room, other than rebuilding the entire front wall, the only significant modification was the addition of a set of custom wood and glass doors, which provide the dual function of isolating noise produced in the machine room, and maintaining bilateral symmetry in the control room. The final monitor system installed at Rocshire is somewhat unique. Actually, it is comprised of several systems, employing TAD, Altec and Tannoy components. Custom enclosures were designed and built, thereby providing each system with nearly optimum placement.

Throughout the reconstruction process at Rocshire, every stage was given more than usual scrutiny. The end result is a state-of-the-art facility that performs as well, if not better, than most studios. And the second studio being planned to take the place of the departed noise-making neighbor should be even better. ■■■

THE GENESIS OF A RECORDING STUDIO
Rocshire's Emphasis on Creativity and Sound Quality

Rocshire Recording's origins can be traced back to a small recording and rehearsal facility. From an inauspicious beginning as an eight-track studio, the facility has expanded virtually overnight to become a parent company incorporating several diverse elements of the entertainment field. Located in Anaheim, California (deliberately based just outside the crowded Hollywood/Los Angeles studio scene), separate divisions are dedicated to the Rocshire and High Velocity labels, artist management, production, and concert sound and lighting. In addition to a state-of-the-art audio recording studio (described in greater detail in the accompanying article) Rocshire features a large room suitable for video shoots, as well as for recording in a live setting, plus a fully equipped remote truck.

Rocshire's staff are no strangers to the music and recording industry. Company president Gary Davis' background is in promotion with such companies as Warner Brothers and Capitol; most recently he held a vice president position at Motown. Vice president Lester Claypool has gained substantial engineering experience through various independent projects, owning his own studio, and as a first engineer for ABC Records.

According to Claypool, who oversaw Lakeside Associates' redesign and construction of the main studio, one of his goals in the design and choice of equipment was to have "a studio that has more to offer than just a shopping list of studio gear." He considers a room's acoustic qualities, various engineer's experience and expertise, along with custom tailoring of equipment, to be more important than a studio's choice of electronics.

LESTER CLAYPOOL

At a time when few record labels own their own in-house facilities, Rocshire's studio remains at the center of its operation, because, according to Claypool, "As a young and developing record company, we find having our own studio allows us to directly supervise our projects, and to control our costs."

A Creative Tool

Claypool views the studio more as a tool to serve Rocshire's artists than as an independent business enterprise. "We produce our own product here," he explains. "That makes us a completely different studio in many ways. Our point of view has always been from the creative side, rather than looking at the music as just an end product. With the high cost of session time, sometimes only the most successful bands can afford a lot of time in the studio. We want our

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Interior of Rocshire's API/3M-equipped remote recording truck.

A separate synthesizer room enable sounds and patches to be set up before session time.

artists to work in a creative environment, without worrying about time and money."

The selection of equipment and design of the label's in-house studio reflects Claypool's recording experience and philosophy. An example is the monitoring system that allows a producer and artists to select from three distinct loudspeaker systems. The idea, he says, was to enable producers to either compare or choose several monitor "sounds," thereby eliminating or minimizing the need to make use of outboard speakers. "It's always seemed funny to me that studios go to the trouble of installing the latest, most expensive, most perfectly aligned monitors, and end up bringing in auxiliary speakers to please the producer or artists. This system helps us please a wider clientele."

The present monitoring system was designed by Claypool with assistance from Ron Becker of Alan Labs, Los Angeles, designers of the Cybersonics disk cutting system. The first system is a UREI 813 Time-Aligned™, utilizing a new horn; the second, coaxial Tannoy SRM-12B units; and the third comprises of a TAD TD-2001 HF driver and a TL-1601 low-frequency driver. All three systems utilize custom, time-compensated crossovers. A TAD TL-1602 15-inch subwoofer can be used in conjunction with any of the three systems. Stax electrostatic headphones and auxiliary outputs are available for those who still prefer to use other monitors. Any of the various monitoring options can be easily selected at the console.

Calypool says that he wanted a quality of sound at Rocshire that was not defined simply by good technical specifications. In particular, the monitoring system should be capable of reproducing the nuances and detail of sound required for audiophile standards. "My goal was to have audiophile sound quality combined with the power and dynamics needed for monitoring," he emphasizes. "The amplifiers, crossover, and Litz wiring all come from an audiophile perspective."

The monitoring system is bi-amplified with Conrad-Johnson MV45A tube amplifiers at the top end, for "inner deatil and musicality," and Soundcraftsmen solid-state amps for the low-end drivers.

The balance of Rocshire's equipment also reflects Claypool's recording philosophy. "I've tried to get a blend of the best of the old and the new," he offers. Making available vintage tube microphones, tube limiters, and tube equalizers neatly balances with Claypool's desire to be "the first on the block with the best new equipment." In several cases Rocshire has indeed been the first to receive new equipment, and has often acted in a consulting role to various manufacturers. For example, the very first Neve 8128 board was installed in Rocshire's studio in late 1982, after being used by the company for demonstration during the Fall AES Convention in Anaheim. Claypool calls this board "utter simplicity," and has found its sound to be extraordinarily transparent.

Synthesizer Room

The most recent addition to Rocshire's main studio is a separate room devoted exclusively to synthesizers. Included are systems by Oberheim (OBX-A, OBX-8, DMX Drum Computer, and DSX Sequencer); E-mu Systems (full eight-voice Emulator with analog interface capability); Simmons (SDS-6 Sequencer, SDS-5 systems for pad drums, and the new TLS Trigger System for use with a standard drum kit); Roland (Jupiter-6, Juno-60, and Juno-6); Moog (Prodigy, Minimoog, and Memorymoog); Sequential Circuits (Prophet-5); and Yamaha (DX-9, DX-7, and GS-1 digital system). Each synthesizer is computer controlled by an Apple II, utilizing custom software written by staff member Bill Boydstun. Boydstun was instrumental in developing the concept of a separate synthesizer room, and Rocshire's emphasis on artists using synthesizers as an integral part of the recording process.

... continued on page 107 —

Claypool and Boydstun have found that a most useful application for the synth room, particularly for artists who are still working on their material, is the ability to pre-program the electronic instruments before a session. "We can not only pre-set the voicings of the instruments, but the rhythm tracks and even arrangements can be set before entering the studio," Boydstun explains.

In this way artists can experiment and work out their arrangements outside of the recording studio, where time is less costly. "I saw how we could get away from the extreme costs of studio time by getting everything worked out exactly on the synthesizers before we went into the studio," Boydstun continues. "From a business standpoint, the use of synthesizers eliminates variables, including human ones. When appropriate for the job, they can save both time and money."

According to Boydstun, a new generation of interface devices allows for complex matching of synthesizer parts required by a piece of music, either with other synthesizer parts, or with previously recorded tracks. New clock and interface devices, such as Garfield Electronics' Dr. Click unit, can lock onto the rhythm pattern recorded on tape, or a programmed electronic instrument, and then synchronize the new synthesized parts to that pattern, even if it is irregular.

"Everything today is being built around the drum tracks, be they acoustic, Simmons-type synthesized, or drum machines," he says. "This lends itself to easy and precise tracking of rhythm patterns."

Boydstun finds that the ability to sync various parts together is extremely valuable while working with people that may already have prepared a rough demo, and wish to develop it further without having to begin from scratch. The new generation of synthesizers and drum machine allows for a human quality in the programming of electronic instruments. "The memory of the new equipment eliminates the machine-sound of previous synthesizers and sequencers," he says. Exact rhythm patterns played by the musician becomes the reference for the other electronic instruments, rather than a mechanical sounding, pre-programmed pattern.

Another use for Rocshire's synthesizer room is in commercial advertising production. Boydstun and Claypool have found that synthesizers can be ideal for use in the strict time formats of 30- and 60-second commercials, because of the precision with which they can be programmed, and the practicaly limitless variety of sounds they offer. Through advance programming, production costs can be minimized, which in turn reduces the amount of studio time necessary, and simplifies the task of editing.

Video/Live-Performance Stage

The newest full studio room at Rocshire is a warehouse-sized space measuring 60 by 50 by 22 feet, with complete facilities for live performance, including stage and lighting. Currently, the room is being used for both video production, and as a way of avoiding the tightly controlled environment of the typical studio. Claypool says that this room "eliminates the fishbowl effect" common to studios, where musicians are being observed at close range by people in the control room. "I think that many artists may feel more comfortable and give a better performance in an environment more like the one where they normally play," he offers.

The shooting of video pieces for Rocshire artists has taken place in the room, along with production of a television series distributed overseas, entitled America in Concert.

Rocshire's other new acquisition is a mobile recording vehicle with complete video synchronization facilities. Claypool says that he purchased the truck to expand the capability of the original studio, as well to provide Rocshire with the ability to handle remote recording dates.

In addition to serving as a control room for the large audio/video studio, the truck doubles as a second remix room. "This is the best mixing truck I've ever heard," Claypool offers. "It has the most accurate monitoring system I've heard in a mobile, especially at the low-end." He points out that, because of the monitoring accuracy, a recent Rocshire album by Maxine Watta that had been mixed in the mobile required no equalization during disk mastering.

After an active first full year of operation, Claypool is quite optimistic about Rocshire's future. With the addition of the synthesizer room, the new audio/video studio, and the mobile vehicle, plus the development of Rocshire's artist roster, Claypool, Boydstun, and Rocshire's studio facilities should remain busy for years to come. □□□

R-e p 108 ▢ December 1983

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COMPUTER CONTROLLED SYNTHESIZERS
IN THE STUDIO

by Paul D. Lehrman

Years ago, it was easy. You wanted a synthesizer, you went to your local music shop and bought one. Whatever you needed it for on a session — lead lines, strings, sound effects — there was a model out there that was right. Most of the time, of course, the thing sat over in the corner behind the Wurlitzer electric piano — waiting for those occasional clients who wanted to use it, and didn't have their own.

But things changed as manufacturers began to offer increasingly sophisticated and complex machines. Although many of them were supposed to be "user friendly," in most cases quite a bit of practice was required to get to know them, and take best advantage of their many features. For studios, it wasn't easy to keep up — it was becoming increasingly impractical to keep on hand every piece of equipment that a client might need, and even more impractical to let staff spend valuable time learning how to make them do their stuff.

Now things are changing again. With the introduction of computer-based synthesizers, the functions and capabilities of many or all of those earlier units — plus, in many cases, the functions of a lot of other studio equipment — have been integrated into self-contained stand-alone units. "Sequencers" have evolved into "note files," with polyphonic keyboard memories equipped for both real-time and off-line recording and editing. Analog-to-digital conver-

ters are no longer used just for sound sampling or "emulation," but can now be backed up with huge amounts of digital storage, allowing them to function almost as true digital recorders, complete with multitrack recording, mixing, and effects.

Of course, by no stretch of the imagination can any of these machines be called cheap. Even though they often cost less than their equivalent in single-task synths and effects units, by and large they remain beyond the reach of the majority of professional musicians. But many studios and production houses, who are often better able to afford them, have been investigating computer synthesizers — not only as nifty toys (which they certainly are), but as ways to streamline the work process, increase flexibility, and draw clients.

Unfortunately, what with new product, it seems, coming on the market (or at least being announced) almost daily, it's still not easy to keep up with what's available. Even for a specialist, staying on top of all of the new developments can be a full-time gig. Hardware manufacturers are born and die, and those that flourish are continuously modifying and upgrading their product. Many spectacular machines you may have seen or heard about have actually been custom-made for one performer or studio, and are unsuitable for mass production, or they exist only in prototype form or on the drawing board, never to become commercially available.

What follows in this article is a brief

survey of some of the more practical computer synthesizers now (or soon to be) on the market, covering a fairly wide price range, and which we hope will be of help to the producer and studio owner. Two caveats are in order: *R-e/p* is not a musician's magazine, and so detailed discussions and subjective judgments of the musical abilities of the systems are best found elsewhere — in this space we are putting the emphasis on how the units relate to the needs of the recording industry. Also, as in many areas within the electronics industry, there is a lot of "R&D by rumor" floating around synthesizer circles, and manufacturers have gotten into the habit of being very tight-lipped about future developments. The author has made a strong effort not to speculate, which unfortunately means that some of the information herein will be rendered obsolete between deadline and issue date. But, of course, that's one of the things that makes this field so exciting.

NED SYNCLAVIER II

The Synclavier, made by New England Digital of White River Junction, Vermont, is probably the best-known computer synthesizer in the U.S. It was first introduced in 1977, and has gone through several major updates. The basic unit (now known as the Synclavier II) is available with from eight to 32 digitally-produced voices, at a price ranging from $16,650 to $31,650. Up to 128 voices are available on special order. The system uses proprietary digital oscillators (two per voice), controlled by a custom computer, whose operating system is written in the XPL programming language.

The nature of each voice (called, very misleadingly, "partial timbers") is determined by balancing 24 harmonics, describing a six-parameter envelope for each, an overall envelope, and vibrato characteristics (waveform, speed, depth, and envelope), and choosing from among several other functions including four portamento schemes, chorusing, and FM harmonic generation.

Up to 32 voices can be triggered with each keystroke. The five-octave keyboard also features up to eight splits, and automatic repeat and arpeggiation. Optional foot pedals and switches can be plugged in to address many of the effects. The octave ratio of the keyboard can be modified, and non-equal scale settings are available. Another option provides stereo outputs.

The keyboard unit (called the "Calvier") includes a panel of 128 buttons, each of which addresses a control function. Parameters are adjusted by a single large knob and a four-digit LED readout. Besides the formidable real-time capabilities of the Synclavier, there is also on board a 16-track "recorder," with capacity for up to 32 voices on each track, and full editing facilities, includ-

SYNTHESIZERS IN THE STUDIO

ing solo, pause, track bouncing, independently-variable pitch and speed, 16 independent loops, transposition, and sync to an external drum machine or tape recorder. An "external clock in" port reads pulses to trigger the unit's sequencer; any type of discrete pulse, including the beat of a live drummer, can be used. "External sync in" is driven by a 50-Hz sync pulse that can be laid on tape or film. An option to read SMPTE timecode is currently under development.

All functions of the machine can be stored on 5¼-inch or 8-inch floppy disks, or for more room and faster access, on a Winchester hard-disk system can be backed up with a Kennedy digital tape-drive system.

Synclavier's Terminal Support Option (about $7,000, including a DEC VT100 graphics terminal with typewriter keyboard, an extra floppy-disk drive, and software licenses), introduced in 1980, extends the power of the system considerably. It includes a music language called Script that allows for precise off-line synthesis and editing from the terminal keyboard. The CRT displays each track on a separate musical staff, complete with key and time signatures and tempo markings. (Compositions entered on the piano keyboard can be converted into Script files, as well.) While in this mode, musical timings — start, stop, speed — can be entered, allowing precise synchronization to audio and videotape recorders or mag dubbers. An adjustable "Frames Per Second" feature allows the user to specify timing events in frame numbers.

The option also provides real-time display of all synthesizer functions on the terminal, either in graphic or alphanumeric format. A third program allows the adventurous composer to write his or her own composition programs, using a language called MAX, which is an extended version of XPL.

Scores created with the terminal can be printed out on a Prism dot-matrix printer in standard musical notation, complete with dynamic and tempo markings and alphanumeric cues, in a variety of score functions.

Then there is the Sample-to-Disk option, introduced in 1982. Using 16-bit analog-to-digital and D/A convertors at a sampling rate of up to 50 kHz, real sounds can be entered, modified and edited (using digital filters), and then played back either right from the disk or with the keyboard. The keyboard can trigger as many as 12 separate sounds, effectively making the system a real-time sound-effects library. In its simplest form, using a single five-Mbyte Winchester disk, the system can store 50 seconds of sound; with eight 40-Mbyte disks on line, the capacity goes up to 54

minutes.

A four-voice polyphonic version of the sampling program is planned for release early in 1984, followed by an "N-voice" version, which won't use Winchester disks for storage, but will require something bigger and faster — although what that device will be has not yet been determined. Prices will be substantial. The company is also working on a method of dumping the output signal directly to a digital recorder, with no intervening analog process.

The latest option is the Guitar Interface, which takes advantage of a device built into the Roland GR Series of guitars for reading pickup output. Synclavier's option converts the pitches to digital periods (*not* voltages), and lets the guitar take over the functions of the keyboard: it triggers and tracks all of the notes, and can even read dynamics and pitch-bend. Synthesized and straight guitar sounds can be mixed in the Synclavier's output. A very convincing demonstration of the system was given at the recent New York AES Convention last October by Pat Metheny.

Obviously, the Synclavier is an extremely versatile unit, with applications that cover the entire recording industry, but with all the bells and whistles it becomes somewhat expensive; a full-blown system can easily top $100,000. In its simplest form, however, the instrument is within the reach of many studios. Fully loaded, the Synclavier is a truly formidable production tool, although for now only the largest and best-capitalized multimedia houses can really afford it.

FAIRLIGHT CMI

The Australian-made Fairlight CMI (Computer Musical Instrument) has been around in one form or another since 1975, when it was introduced as the Qasar. In many ways, the Synclavier and the Fairlight are moving towards each other — whereas the former started as a synthesizer and recently added sampling, the latter gained notoriety as an emulation


NED Synclavier II, with optional guitar interface shown below.

machine and has since improved its manipulation and synthesis capabilities. The basic CMI unit ($22,000 for four voices, $27,750 for eight) consists of a six-octave touch-sensitive piano keyboard, a digital processor with two 8-inch floppy-disk drives and a built-in 20-watt power amplifier, an alphanumeric keyboard with a 15-inch video monitor, and a light pen.

The Fairlight sampling software is designed to take maximum advantage of available memory. The input sampling rate is continuously variable from 2.1 kHz to 30.2 kHz (output sampling goes as high as 150 kHz), and the sampling time varies inversely from 0.5 to 8 seconds. The computer chooses an input

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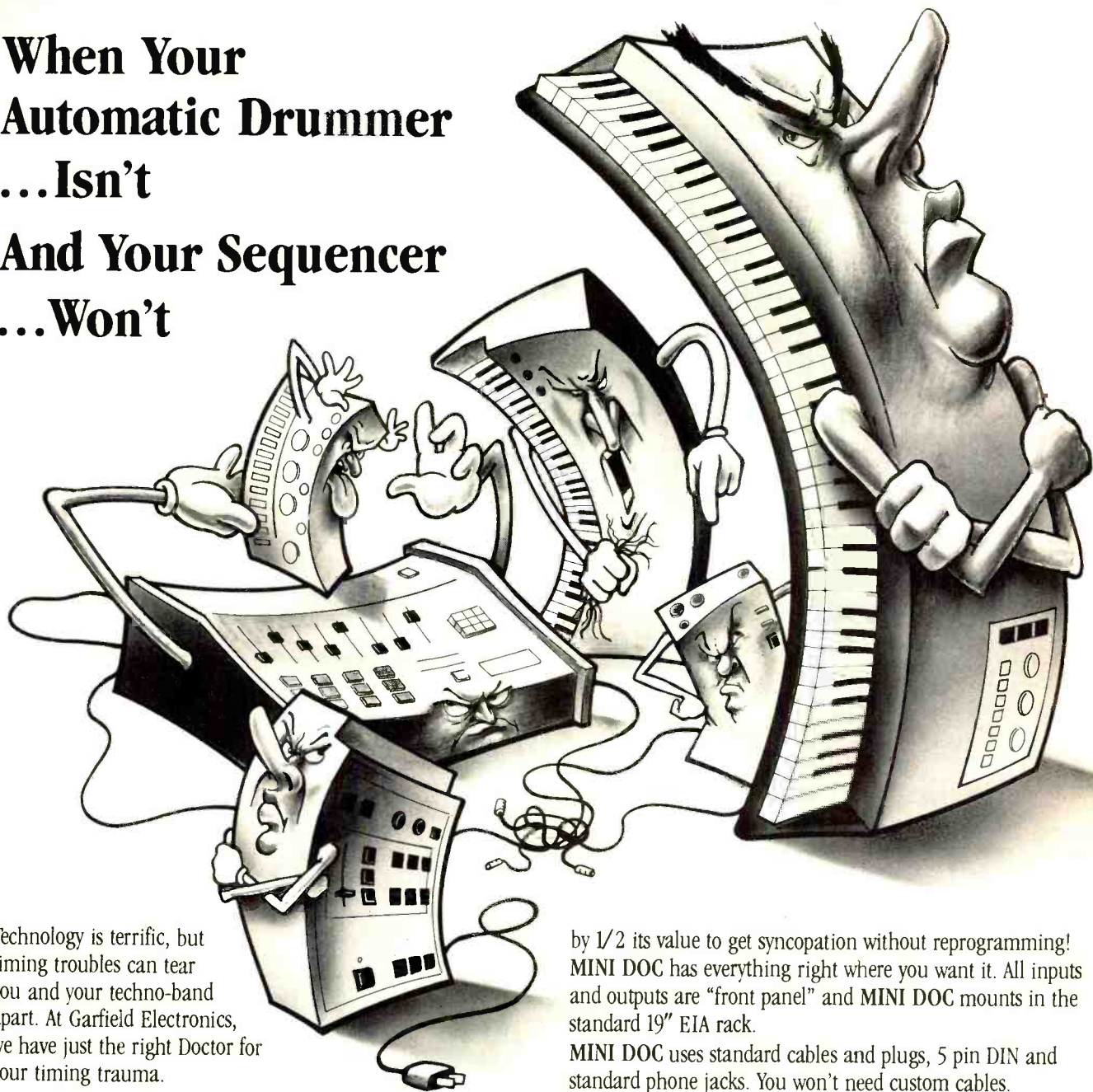
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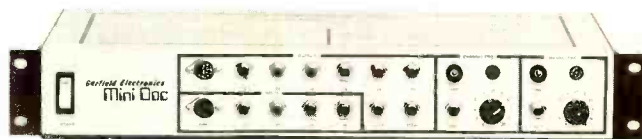
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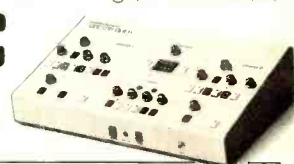
SYNTHESIZERS IN THE STUDIO

sampling rate that is an integral multiple of the sound's fundamental frequency, so that the wave can be "looped" (sustained) indefinitely without glitches. Since each voice uses its own hardware card, multiple sampling rates are used in the polyphonic mode. Resolution is only eight bits, but various schemes, including bit-packing and word-ganging, are used to provide a signal-to-noise ratio of 85 dB.

Once a sound is loaded into the Fairlight's memory (the instrument is furnished with a library of some 450 sounds — about 20 instruments fit on each disk), it can be manipulated in a variety of ways. Each sound can be assigned various control parameters, such as volume, envelope, pitch, vibrato, sustain (which loops the sample), and portamento, which can then be accessed automatically or with foot pedals, faders, or the touch-sensitivity function on the piano keyboard.

The waveforms themselves can be turned around on themselves or merged and balanced with other sounds to create new waveforms, and can even be redrawn with the light pen. The light pen also allows mapping of 32 harmonics, each with its own envelope.

Once the waves are completed, they can be played in real-time on the main keyboard or on a slave keyboard. An internal polyphonic 50,000-note sequencer can be loaded from the piano keyboard, or by using the typewriter keyboard and the included Music Composition Language software. Lines entered in real time can be edited and corrected off-line.

A new development (now part of the main package) is the Real-Time Composer. This program can arrange up to eight voices in 256 patterns, and access eight of them at a time to be performed, edited, or looped, all while the machine is playing.

For more complex arrangements, the Fairlight can be triggered by an external sync pulse, so that it can then dump note files in sync onto a multitrack tape deck. A click track with separate output is provided. Each voice card has its own balanced audio output so that voices can be processed (EQ, delay, reverb, etc.) individually.

Options available include a SMPTE interface card, various printer and plotter packages for recording screen graphics or transcribing note files, and an Analog Interface Controller for reading data from external analog devices, such as a guitar or analog synthesizer. A MIDI interface is also being developed.

The Fairlight's chief advantage seems to be that it is relatively fast, which is partially due to the new composition software, but also since the basic sonic materials are real sounds, the time-consuming procedures of analyzing and reconstructing waves can be avoided. The disadvantages of sampling a single sound and replaying it at various pitches (at the extremes, such sounds never sound real) are at least partially overcome by the system's waveform-merging capabilities. Although the Fairlight is only capable of producing eight voices at a time, it is flexible and fast enough to be able to produce full-sounding orchestrations quickly in a studio environment.

FAIRLIGHT CMI

MCLEYVIER/IMP

Developed in Toronto by a team led by musician and computer designer David McLey, and considered by many a likely candidate for inclusion in the "supersynth trinity" after its introduction at the 1981 New York AES Convention, the McLeyvier unfortuntely soon dropped from sight. There were several possible reasons for this initial failure, but the one that seems most likely is that, while the company had developed a superb music-notation program using a high-speed graphics plotter, the *sounds* that the $50,000 digitally-controlled analog machine made were reportedly not very good.

Although the unit is still available in highly limited quantities (one disgruntled member of the original team claims that it comes without any support, but this appears to be rather an overstatement), in the meantime a new team, headed by New York electronic-music composer and instrument designer Laurie Spiegel, is taking off from the original McLeyvier to build a totally new machine, designed for the ultimate in compositional and orchestrational flexibility. It is tentatively named the IMP, standing for Interactive Music Processor.

The IMP uses a keyboard, equipped with a new optical velocity-sensing feature designed for better accuracy and reliability than mechanically-based

MCLEYVIER/IMP

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PPG WAVE AND WAVETERM

systems. It contains 16 individually-programmable digital oscillators, with room for expansion. Audio sampling is handled by the oscillator boards, using a 16-bit First-In-First-Out scheme with a 48-kHz sampling rate. Waves are stored in extra-large 2-Kbyte tables. Also on board are eight-bit A-to-D convertors for custom interfacing of analog controllers, such as joysticks or pedals.

For now, the heart of the system is the original McLey-modified DEC 16-bit PDP-11/23 minicomputer, but Spiegel eventually hopes to put on line a Motorola 68000, as well as several other processors that will make the system more modular and efficient. Storage is handled by a 10-Megabyte Winchester disk. "Everything you have ever done with the machine is on-line in real time,' Spiegel says. A SMPTE timecode reader and generator are built in.

The original scoring program, whose high quality caused quite a stir at AES, will remain available, but other options are being developed, including user-programmable graphics that will allow truly exotic notation and editing schemes.

According to Spiegel, the most important feature of the IMP is that it is being designed around a brand-new programming language, somewhat similar in structure to FORTH or BASIC, that will allow the user to customize to a great degree the synthesis and notation programs. For example, FM synthesis, waveshaping, sampling, and digital filtering might be available for different voices all at the same time. The original McLeyvier had 180 computer commands; the IMP starts from there and allows the user to create an infinite number of new commands that can be used for abstract and algorithmic composition, as well as to redefine performance and recording functions.

"It's like a word processor," Spiegel offers. "It can manipulate musically meaningful units, not just individual events. It can work like a multitrack tape recorder, but it doesn't have to; a composer can play a series of chords, and then go back and re-orchestrate each note individually." In addition, any programming structures for synthesis, editing, or notation that a user designs can be stored on a floppy disk and loaded into another machine at another location.

It's going to be a while before you can see a working IMP: "Something demonstrable will be available by February," says Spiegel, and a commercial unit should be out within a month or two after that date. Because the price of hardware has fallen so much since the original McLeyvier was introduced, the new unit should come in at a much more resonable cost than its predecessor; Spiegel predicts it will run between $25,000 and $30,000.

PPG

The latest entry in the do-it-all synthesizer sweepstakes is a German machine, PPG, that has been available in America only since last March. It is a hybrid, combining digital synthesis with digitally-controlled analog filtering, and it offers many of the features of the pure digital machines at a reasonable price.

The PPG comes in two parts: the Wave 2.2 ($7,950), which is a stand-alone five-octave, 16-voice keyboard synthesizer, and the Waveterm ($8,950), an add-on computer.

The keyboard unit has nearly 2,000 waveforms on board, organized into 30 tables of 64 each. Generally, each synthesizer voice uses two oscillators but, if desired, all 64 waves in a table can be accessed, in any order, with each keystroke. Parameters can be set with a numeric keypad, and are shown on an 80-character LCD display.

Three sets of front-panel envelope controls double as filter controls, and can even serve as a mixing console for the built-in 1,000-note, eight-track sequencer. The mix can be automated as well, with no external equipment. The sequencing program allows lines to be entered — one at a time, or polyphonically — and edited. In addition, the voice parameters can be adjusted after recording, and the adjustments stored in memory.

The Waveterm is a dual-8-inch floppy-disk drive unit with video screen, based around a 6809 processor. The programming is divided into several "pages." One page creates waveforms by additive synthesis of the first 32 harmonics, either over a sine wave or over a

December 1983 □ R-e/p 117
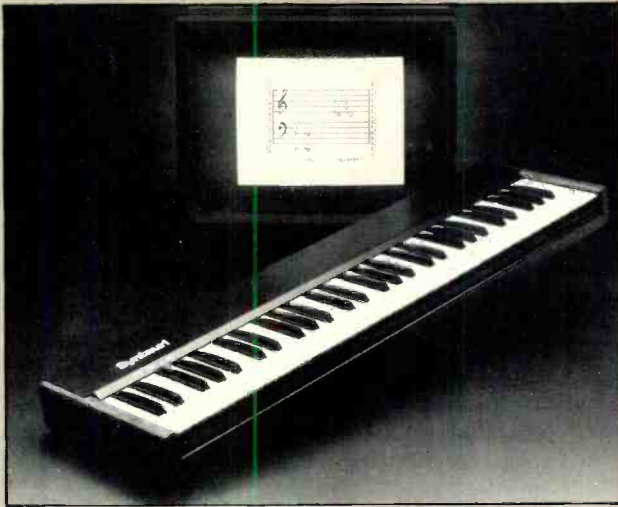
ALPHASYNTAURI WITH COMPANION APPLE II

PASSPORT SOUNDCHASER/ APPLE II COMBINATION

at a time, and *will* interface with MIDI, is being developed.

ALPHASYNTAURI

At the low end of the price spectrum is the AlphaSyntauri, made by Syntauri Corporation of Los Altos, CA. Unlike the Chroma, Synclavier, and PPG, the Alpha is totally dependent on its computer: it uses an Apple II for storage, manipulation, and synthesis. The heart of the hardware is a pair of circuit cards known as the Mountain Computer Music System, which plug into expansion slots on the Apple. The cards contain 16 digitally-controlled oscillators, feeding stereo high-level unbalanced outputs.

The Alpha comes with either a four-octave or a velocity-sensitive five-octave Pratt-Reed keyboard, which also plugs into the Apple. The accompanying software breaks down into two main modules, each on its own disk: Alpha Plus, and Metatrak.

Alpha Plus is the instrument-generation system. Each instrument occupies two independently-controlled oscillators. Waveforms can be generated with any of several programs, among them Quickwave, which balan-

For additional information circle #77

ces 16 harmonics above a base wave; Draw Wave, which allows you to graphically design waveforms with a pair of game paddles, or using a 256×256 point or vector map; and Wave, which lets you mix four basic waves, adding and balancing an unlimited number of harmonics as you do so; as well as programs for Hammond B3-organ simulation and highly-precise pulse generation. Envelopes can be defined for the finished waves, with six parameters on one oscillator and four on the other, and the oscillators can be balanced and assigned channels.

The completed waves and envelopes are then arranged into "preset masters" of 10 instruments, which are stored on floppy disk (each disk has a capacity of about a dozen preset masters), and then called up for use in Metatrack mode. (A full disk of preset masters is furnished with the package.) Metatrak allows real-time entry of polyphonic voices from the piano keyboard, which can then be overdubbed for up to 16 tracks. Several editing functions are available during the overdubbing process. Different instruments can be assigned to each track so that re-orchestrations are possible.

Keyboard splits can accommodate up to eight instruments, and multiple instruments can be played with a single keystroke. Several special effects are available, including AM and FM vibrato, real-time transposition, loop-

KURZWEIL 250 PROTOTYPE

ing, etc.

There are facilities for synchronizing the playback to a drum machine and to a multitrack tape deck. This last is accomplished by the system's writing a digital "start pulse" on the first track, which it can then recognize and lock up to on subsequent passes. Optional software packages provide education programs and a music-notation printing program, which is rather slow and

hardly professional quality (especially when compared with the McLeyvier), but nonetheless quite readable.

For about $2,000 (plus the cost of the Apple II computer), the Alpha does quite a bit, but its relatively low price does mitigate a few drawbacks. The system does not, as yet, support real-sound sampling in any form, nor can it produce really convincing white noise; as a result, many percussive sounds are dif-

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ficult, if not impossible, to synthesize (hence the drum-machine interface). The most significant disadvantage is the sound quality, which is limited by two factors: the eight-bit format of the Apple, which can only provide a certain level of dynamic resolution and freedom from noise; and the Mountain cards. With a sampling rate of 32 kHz, the card's frequency response should be good to 16 kHz, but it is all too easy to drive the on-board oscillators into serious aliasing, which limits the amount of high frequencies you can generate safely. There are indications that Syntauri will soon be announcing a solution to this problem.

In practical terms, however, these limitations will only become apparent when you try to perform complex arrangements on an unaccompanied Alpha, and even then you can get away with a lot. One of the major advantages of the system (besides cost) is that, being software-based, it is easily upgradable as the company developes new programs, or improves existing ones.

PASSPORT SOUNDCHASER

Another system along lines very similar to the AlphaSyntauri is the Soundchaser, from Passport Designs of Half Mood Bay, CA — it also uses an Apple computer and the Mountain oscillator cards. One major difference is that the system is sold on a more modular basis: the basic package ($1,190) includes a four-octave keyboard, the Mountain cards, 50 preset sounds, waveform-generation programs, and a four-track recording system with a capacity of about 4,400 notes. Sixteen-track recording (called "Turbo-Traks") costs another $250, and also provides somewhat expanded capacity and a bank of special effects, including a "digital filter." This is not like a dynamic analog filter, which changes the sound as a note progresses, but rather it is a computer subroutine that helps to minimize aliasing and other digital noise by allowing you to specify a low-pass cutoff frequency for each wave during the wavebuilding process.

Other packages include a system re-tuner, drum-machine sync, and a four-track transcription/printing system. One Soundchaser program not yet offered by Syntauri is an off-line composer/editor, useful for musicians with limited keyboard chops.

Like Syntauri, Passport Designs is constantly modifying its product, as evidenced by a recently-announced upgrade and reconfiguration, scheduled, as we go to press, for December. It will consist of a new five-octave keyboard, bundled with much of the performance and recording software, and a custom-designed oscillator system that will combine the functions of the two Mountain cards, as well as the keyboard and drum-machine interface, on a single circuit board. It is hoped that the

new design will eliminate many of the sonic problems of the Mountain cards.

It's obvious that, although they were developed separately, the Soundchaser and the AlphaSyntauri have turned out quite similar. While the Syntauri holds the edge in flexibility (specifically with regards to waveform-generation, envelope control, and the velocity-sensitive keyboard), the Soudnchaser is a little faster (both the wavemaking and the recording programs are on the single Turbo-Traks disk), and a bit easier for the beginner to use.

Passport Designs is currently turning much of its attention to software for other instruments, particularly MIDI-equipped products. It doesn't plan to implement MIDI on the Soudnchaser,

but rather hopes to become a sort of OEM source of programs for other manufacturers' instruments and computers. Among its first releases will be The MIDI Network, which will allow recording, overdubbing, and editing on several model of synthesizers and drum machines with an Apple II computer.

KURZWEIL 250

Audio engineers are generally a skeptical bunch, but at least one demonstration at New York AES caused nearly everyone to shake their heads in amazement: the Kurzweil 250. Still in prototype form, the machine's release

date is uncertain as of this writing, although it will be soon. The price is up in the air as well, although it has been announced at various times to be anywhere between $7,000 and $10,000.

The Kurzweil is basically an emulation machine but, to this writer at least, it does that task far better than anything available previously. The sound that the company is most proud of is its "grand piano" which, as anyone familiar with synthesis knows, is a fiendishly complicated sound to reproduce.

Although the PA system used during the AES demonstration left a bit to be desired, the quality of the Kurzweil's piano sound was unmistakable.

At the heart of the system are about 60 128Kbyte ROM chips, each of which, depending on the complexity of the sound, contains between one and a dozen "instruments." Because the chips always remain in the machine (this is the only unit discussed here that operates on a single chassis), any of the voices can be called up immediately. Voices can be changed or added by swapping ROM chips, or by inserting videogame-like cartridges. The velocity-sensitive piano-sized keyboard can be split into 88 separate instruments. Up to 12 notes can be sounded simultaneously.

The keyboard is specially designed and weighted to feel like a real acoustic piano. Eight real-time slider controls can be set to adjust any parameters of the sound model. The machine will come from the factory with the grand-piano chips installed, and the purchaser can specify whichever other instruments he would like included. The software also includes provisions for user sampling of sounds up to five seconds in length, with adjustable word length and sampling rate. An option will be available to extend that sample length to 20 seconds.

The on-board computer includes a 7,500-note, 12-track sequencer, with provisions for editing, track bumping, transposing, re-orchestration, and independent speed and pitch change. The system does not use a CRT — instead, information is shown on a two-line-by-24-character LCD display. Provisions are still being developed for attaching an external computer to provide extra memory for sequences and sound files, as well as off-line composition and editing. Trigger-sync inputs and outputs are provided for slaving to and from other synthesizers, and a 12-channel MIDI interface also is included.

LUCASFILM ASP

Although the ultimate music machine will fit into a six-foot high 19-inch rack, it will not be available at your local dealers. It will do everything: synthesis, recording, editing, processing, sweetening, video and film lock and, more than likely, can be taught to make lunch. It's the Lucasfilm's Audio Signal Processor, the result of several years of R&D by the folks who brought you Star Wars — but this ain't science fiction.

When it is released commercially, the ASP (a particularly appropriate acronym) will contain a 32-track digital recorder and synthesizer, but outside of four 300-Megabyte hard-disk drives, it will have no moving parts. As if that weren't revolutionary enough, the system will be useable with any kind of "front end": an audio or film mixing console, a keyboard or touch-pad synthesizer, or even a graphics tablet set up for totally abstract composition.

The technical details you will have to look for elsewhere [in particular, refer to Larry Blake's two-part article on the sound for Return of the Jedi; October and this issue — Ed]. Suffice it to say that the ASP will easily replace any and all gear now used in any kind of audio work (except maybe Nagras and the odd condenser mike), as well as half of the engineers and musicians who now make their living from sound recording. It will be available around Christmas time of 1984. ∎∎∎
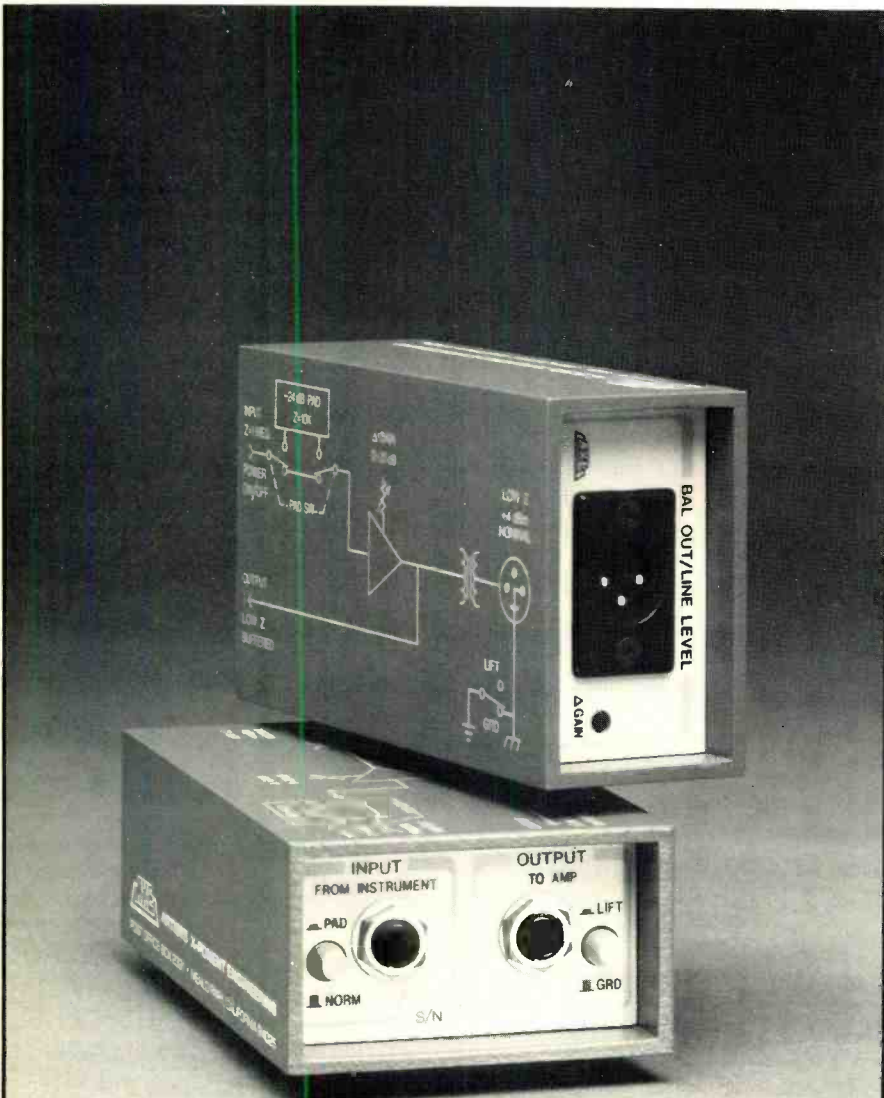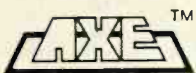
A RECORDING ENGINEER'S GUIDE TO MIDI
The Musical Instrument Digital Interface Explained

by Jeffrey Rona

The producer walks into the control room and sits down. The people from the cartage service have gone, leaving an array of synthesizers, drum machines, a small computer, and several empty styrofoam cups with cigarette butts in them. The engineer finishes calibrating the last multitrack channel as the producer reaches into his Anvil briefcase, and pulls out a 5¼ inch floppy disk. "Let's cut it," he says in assured tones.

No sooner than the red light on the disk drive goes off, than the room is alive with music. Each instrument playing its part in perfect synchronization with each other. The horns from the Roland, strings from a Prophet, and clavinet from a Yamaha. Three and a half minutes later, it stops. "Perfect. Vocals tomorrow, and we mix. See you."

Possible? Well, yes and no. Yes, most of that story is conceivable with MIDI. But no, not for a little while.

MIDI — or Musical Instrument Digital Interface for short — is a new, industry standard for interfacing electronic instruments. It was created over the past two years by members of several of the larger synthesizer manufacturers, and arose from a large public demand that synthesizers of various makers be more compatible with each other. That is, if I buy Brand X synthesizer (for a not-small sum), and later fall in love with and buy Brand Y, how will I be able to use them together at the same time? Several people have spent small fortunes to pursuade synthesizers from different manufacturers to function together from one keyboard. Sequencers, the backbone of more than one hit record, was another concern. If we will make these keyboards able to "talk" with one another, how about making them also controllable from the outside word. The answer to these problems was the creation of MIDI.

Since some form of communications is necessary to carry out a task like this, it was realized that such an interface could be patterned after existing computer communications techniques to allow for eventual universal synthesizer/computer links. Those familiar with computer ASCII code will see big similarities throughout. Performances done on a MIDI keyboard are converted into a special digital code, the MIDI code. This code is then sent out the back of the synthesizer through a five-pin DIN plug marked (logically enough) OUT. Any synthesizer picking up the code through its IN plug can translate the information back into an identical performance using its own sound equipment. Further, there is on most MIDI instruments a THRU plug which allows MIDI to be sent on to a third, fourth or near infinite number of instruments (if the room and budget allow).

It is important to realize that MIDI by itself has no concept of the sounds being produced, or even if it is a synthesizer creating noises at all. It deals strictly with the *operation* of the device; what is being done to it. The code is translated back into the same sequence of actions. It is up to the user to think in terms of sounds, patches, etc.

MIDI has several different configurations, or modes, in which it works. To begin, MIDI has 16 channels over which it can send data. As with cable TV and the like, the channels are sent over the same line, and are "tuned into" at the receiving end. In MIDI's initial mode *(Omni On/Poly)*, any information sent over any channel is received and played by any synthesizer. In this mode, the output of a synthesize could be run into the input of a second one for a master/slave set-up to give a layering effect. All MIDI synthesizers currently available use this mode exclusively. In the *Omni Off/Poly* mode, there is the option of having different synthesizers daisy chained together on one line, and each play a separate channel of music polyphonically. This will be in a second generation of MIDI instruments that have the ability to "tune" to the proper channel. In addition to these two modes there are two *Mono* modes for sending individual note events through the channels. These also will require special hardware considerations from the instruments.

With MIDI any synthesizer can be connected to any other synthesizer regardless of brand, model, number of voices, or any specs. This already solves many problems. But to paraphrase the song, "is that all there is?" The short scenario at the beginning indicates a resounding "no." MIDI can be linked very simply to most any small computer through a MIDI interface. These will be available quite soon from a number of keyboard manufacturers. Such interfaces will allow information coming from the MIDI line to be taken, processed and stored for later retrieval. This has multiple possibilities. Parts can be generated beforehand and synchronized with other tracks. Since each key of a keyboard is nothing more than a number to the computer, it is possible to set up sequences and other cues in the computer, and have live musicians be able to trigger them at will without their hands even leaving the keyboard. This could be useful in both live and recording environments. MIDI interfaces will allow for simple tape sync, along with multiple instrument synchronization. It would be a simple matter if a keyboardist plays a hot solo, but the sound is wrong, to retake the exact same solo with any other patch setting. It will be possible to record fewer tracks, and leave several instruments to be played live into the final mixed master, no matter how many takes are needed. It will be possible, to a limited degree, to use MIDI to assist in automating outboard gear during recording and mixdown (all in sync with the music).

The possibilities with the computer are limited only by imagination and new hardware. MIDI does not need to be restricted to keyboards alone, though it was designed for them. Lexicon, for example, is said to be coming out with MIDI-compatible devices in the future. J.L. Cooper Electronics in Marina del Rey, California, has released an interface for converting MIDI data into Control Voltages (and CV to MIDI), for control over other non-MIDI synthesizers and out-board equipment, or vice versa. As the technology increases, so will the applications. As the

— the author —
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demand for MIDI increases, so will the technology. We will be seeing numerous unique MIDI applications beyond keyboard interfacing in the near future.

It is perhaps unfortunate that MIDI is being so hotly hyped right now, mostly by overly eager sales people. It is a virgin technology which, while full of promise, will not be the near panacea it is being made out to be. As it matures, and fulfills its own precepts, MIDI will be a useful tool for both the musician and musical engineer. We can only use it, try new things with it, and stay in touch with the MIDI manufacturers and programmers to let them know our findings, ideas, and complaints.

AND FOR THE TECHNICALLY MINDED . . .

PROTOCOL AND IMPLEMENTATION OF MIDI VERSION 1.0

The Musical Instrument Digital Interface (MIDI) specification evolved as a cooperative effort by several synthesizer manufacturers to enable the integration of synthesizers, other electronic keyboards, sequencers, drum machines, and home computers into one programmable system. It should be stressed that MIDI is not just an interconnection scheme; it also provides a fully-documented operating protocol, and defined instruction set. Via MIDI, synthesizers can be configured "in series," with instruments played simultaneously or remotely; entire compositions, consisting of monophonic and polyphonic sequences and rhythm, can be played at one touch; a computer terminal can be used for composing, sequence creation, and editing; and video synthesis can be integrated with music synthesis.

To simplify cabling between instruments, the MIDI interface is serial, and operates at 31.25 kBaud asynchronous. This rate is considered a high speed for serial operation — in comparison to the typical RS-232 maximum of 19.2 kBaud —and was chosen to prevent objectionable delays between equipment; the 31.25 kHz clock also can be obtained easily from hardware — for example by dividing a 1 MHz master operating frequency by 32. One serial data byte consists of a start bit, eight data bits (D0 to D7), and a stop bit — for a total of 10 bits transferred in 320 microseconds.

Physically, MIDI appears as two or three jacks on the instrument; the connectors are DIN five-pin (180-degree) female panel mount receptacles. However, the specification does provide that a manufacturer can use XLR connectors, if the company makes available all necessary conversion cables.

Most MIDI instruments will have two jacks marked MIDI OUT and MIDI IN. The transmitter data typically originates in the instrument's UART (see accompanying figure). The interface circuit is a five milliamp current loop, designed especially to prevent the formation of audio ground loops that often develop in complex systems; the output normally is meant to drive only one input. If transmit data is low (0), current flows from Vcc (+5V) through RA, over pin #4 of both connectors, through the opto-isolator, returns over pin #5, then through RE. The opto-isolator output is normally pulled high by RD. However, when current flows through the internal LED, the isolator output switch turns on, grounding V0, thus sending a low to the receiver UART. When data is high, the LED does not light — the receiver UART therefore sees a high. D1 protects the opto-isolator from reverse-polarity currents that may result from transmitter anomalies.

Interconnect cables should not exceed 50 feet, and should be shielded twisted pair, with the shield connected to pin #2 of the 5-pin DIN plugs at both ends. (While the MIDI OUT jack is grounded to the instrument chassis, MIDI IN is not; this allows the cables to provide shielding without creating ground loops.)

The optional third jack, MIDI THRU, provides a direct copy of data coming in MIDI IN, and is included when a manufacturer intends the instrument to operate in a "chain" or "loop" network, as opposed to a "star" network.

Modes and Channels

The first thing to realize about MIDI is that the total control features available still depend on the design of each specific piece of equipment; MIDI does not transcend equipment limitations or differences. Rather, it merely enables them to "communicate" at their "least common" level. For example, specific programmed sounds cannot be transferred directly between different models of synthesizers because of inherent design differences, but keyboard information and program selections can be communicated. One of MIDI's design goals was that it be simple enough to allow connection of any polyphonic synthesizer to any other, or to a sequencer, and at the very least the notes would be correctly played or stored.

Each type of musical equipment has different minimum requirements. For synthesizers, min-

ROLAND'S MIDI-EQUIPPED SYNTHESIZERS INCLUDE THE JUPITER-6 . . .

. . . JX-3P . . .

AND PB-300 RHYTHM PLUS

MIDI INTERFACE CIRCUIT

SHARP PC-900 OPTO-ISOLATOR
D1 IN914
+5V
R$_H$ 270 OHM
5V
Vo
GND
NC 5 2
NC 3
NC 4
NC 1
MIDI IN
R$_H$ 220 OHM
TO UART
+5V
OPTIONAL
R$_A$ 220 OHM
A
R$_C$ 220 OHM
MIDI THRU
NC 5 2
NC 3 NC 4 NC 1
+5V
R$_A$ 220 OHM
FROM UART
A
R$_C$ 220 OHM
5 2
3 4
NC 3 NC 1

Note: Gates "A" are open-collector TTL or LS.
Resistor are 5% tolerance.
HP 6N138 opto-isolator may be used with appropriate component changes.

imal usefulness seems to include remote keyboard control and program switching. While polyphonic sequencers send and receive keyboard data, they may or may not be interested in program changes. Monophonic sequencers can only deal with individual lines, so keyboard data must somehow be different for them. Digital drum units usually don't care about specific keyboard notes, but may need to synchronize to their timing, or to the sequencer, and perhaps react to program changes as well.

While most of these requirements and useful control options can be foreseen, the number of possible interconnections cannot. Therefore, although the MIDI specification says that each transmitter will drive one and only one receiver, provision has been made so that any specific instrument or synthesizer voice on the MIDI bus can be addressed, regardless of the interconnection scheme. This is accomplished by assigning up to 16 *channels* under increasingly complex modes.

Each unit connected to the MIDI bus has separate transmit and receive ports. There are four modes of operation for transmitters and receivers: Omni On/Poly; Omni On/Mono; Omni Off/Poly; and Omni Off/Mono. The standard default mode is Omni On/Poly. The other modes are used in special situations where separate control over synthesizers or even voices within an instrument is required.

Omni On/Poly: Upon power-up, all instruments will be in the default mode of Omni On/Poly. Regardless of system configuration, in this mode the instruments will send all data over channel #1. They can receive Note On/Off events, and all data bytes sent over any channel. The internal hardware for voice assignment acts as it normally does to send voice data to the correct sections of the synthesizer. As a result, Omni On/Poly mode allows any number of instruments to be connected and play in series. A receiver's mode can only be changed by a Mode Select command sent over the current channel(s). If it is not capable of the new Mode, the command is simply ignored. Even though a receiver in the Omni On/Poly mode receives all channels, it will only respond to a Mode Select command in only one channel: the one to which it is assigned.

Omni Off/Poly: While Omni On/Poly allows an instrument to receive data from all channels, in the Omni Off/Poly mode, it will only receive data sent over a single channel. The data can still be polyphonic in nature, but data addressed to other channels is ignored. Internal channel assignment is still handled by the instrument's own hardware in the usual manner. This mode would allow various instruments connected to the same line to play completely separate lines. However, to achieve this the host system would need to be able to produce the independent parts, and send them via the separate channels.

Omni On/Mono: In this mode, commands are received over all channels, but they will only control a single monophonic voice. This allows a polyphonic device to emulate a monophonic one, such as a "lead" synthesizer.

Omni Off/Mono: Voice messages are received monophonically over several channels, each one controlling a single voice of the receiver. The number of actual voices is determined by the third byte of the mono mode command. This mode disrupts, and takes over, the internal channel assignment hardware of the receiver.

Data Format

There are four categories of MIDI data: Channel, System Common, System Real Time, and System Exclusive. Each data category encompasses a number of "status bytes" that define specific commands under that category, and which precede data bytes that specify the exact operation. Status bytes are distinguished from data bytes according to whether the most-significant (MS) bit is set (1=status) or reset (0=data). The status bytes under each category are defined below. (It should be noted that any data sets which are sent successively under the same status can be sent without a status byte until a different status byte is needed.)

Channel information performs most of the routine work. Commands are addressed to specific channels by a four-bit number that is encoded into the status byte. The associated data bytes can identify keys going down (on) and up (off); their on or off velocities; and pressure or "after-touch" (on keyboards so equipped). System Common and Real Time information is intended for all channels in a system: System Common information identifies song selections, measure numbers and Tune Request for all

YAMAHA DX7

MEMORYMOOG
AND OUTPUT
PANEL (RIGHT)
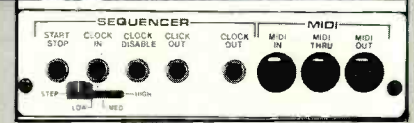
units; Real Time information is used for synchronizing everything (perhaps to a master sequencer), and for performing Reset functions to the system. Therefore, Channel and System Common information can be interrupted by System Real Time information. System Exclusive information allows the exchange of data that can be formatted as the manufacturer wishes; only devices which recognize the manufacturer's format will attend the exchange. Reset simply initializes all equipment to power-on condition.

The four categories are ordered below according to their utility (note that, for example, 9nH signifies a general hexadecimal, base-16 number, while 1001nnnn is the binary representation of an 8-bit byte):

Channel: The most significant four bits of each Channel status byte define the command, while the least significant four bits identify the effective channel.

9nH: Note On Event, comprising three, 8-bit bytes: 1001nnnn + 0kkkkkkk + 0vvvvvvv, where *nnnn* is the channel code, 0-15, corresponding to channel numbers 1 thru 16; *kkkkkkk* the key number (0-127 for all keyboards, middle C=60, and all C key numbers are multiples of 12); and *vvvvvvv* the Key-on velocity, 0-127 (1=softest, 127=loudest).
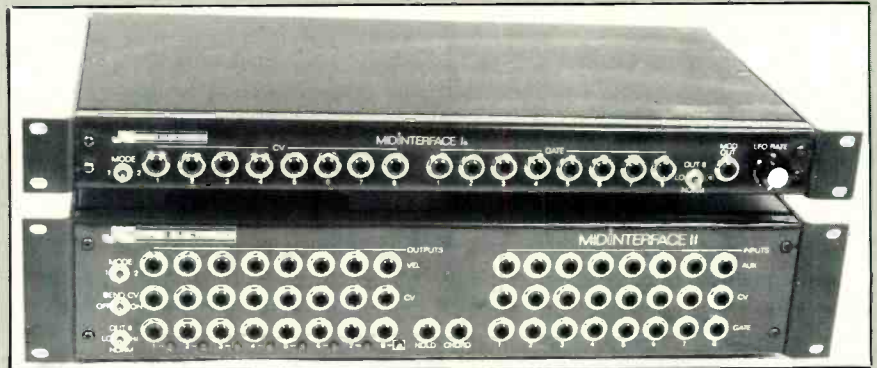
J.L. Cooper Electronics manufactures two MIDI-to-Control Voltage devices, illustrated above. The MidInterface Iin converts up to eight gates and CVs into MIDI information; for example, to attach the CV and Gate outs of an Oberheim DSX to drive a MIDI-based synthesizer. The MidInterface Iout converts MIDI information into up to eight CVs and gates. Options such as cyclical (Oberheim) or recall (Prophet) assignments, Low-note/High-note output, Pitch bend output, Modulation output, Mono/Omni Modes, and high/low active gates are available. The MidInterface II is a bidirectional unit that converts MIDI/CV in both directions, including velocity information. Also available is an E-mu Systems Emulator to MIDI modification; the Emulator must be equipped with RS-232 interface and MIDI software. Full details from: J.L. Cooper Electronics, 2800 South Washington Blvd., Marina Del Rey, CA 90292. (213) 827-4884.

8nH: Note Off Event, comprising three, 8-bit bytes: 1000nnnn + 0kkkkkkk + 0vvvvvvv, where *vvvvvvv* is the Key-off (release) velocity, 0-127.

AnH: Polyphonic Key Pressure, comprising three, 8-bit bytes: 1010nnnn + 0kkkkkkk + 0vvvvvvv, where *vvvvvvv* is the Pressure/After-touch value, 0-127.

BnH: Control Change, comprising three, eight-bit bytes: 1011nnnn + 0ccccccc + 0vvvvvvv, where *ccccccc* is the Control address, 0-127 currently, the controllers are not specifically defined; continuous controllers, including the Pitch Bender, are divided into Most and Least Significant Bytes. If only 7 bits of resolution are needed for a specific controller, only the MSB is sent. If more resolution is needed, then both are sent, first the MSB, then the LSB.); and *vvvvvvv* the Control value, 0-127 (pitch benders should range from 0-127, with 64 being center —no pitch bend; other controllers will range from 0=minimum to 127=maximum). The manufacturer may assign the logical controllers to physical ones as needed. A controller

allocation table must be provided in the user's manual.

CnH: Program Change, comprising two, eight-bit bytes: 1100nnnn + 0ppppppp, where *ppppppp* is the Program number, 0-127.

DnH: Channel Pressure, comprising two, eight-bit bytes: 1101nnnn + 0vvvvvvv, where *vvvvvvv* is the Channel pressure/after-touch amount, 0-127. (For mono mode the channel, rather than key, is identified.)

EnH: Pitch Wheel Change, comprising three eight-bit bytes: 1110nnnn + 0vvvvvvv + 0vvvvvvv, where *vvvvvvv* is the value of the pitch bender in Least Significant and most Significant bytes. The sensitivity of the wheel will be determined by the receiver. The center position of the wheel (no change) is 2000H, which would be sent as EnH-00H-40H.

System Exclusive: A format has been defined for System Exclusive information, consisting of a two-byte preamble, the data itself, and a one-byte end code. The purpose of this format is to provide for the transmission of data that may be

SEQUENTIAL CIRCUITS' MODEL 64 SEQUENCER, PROPHET 600 AND PROPHET T-8

useful to any two instruments from one manufacturer, but uninterpretable to other MIDI-bussed devices. An "End Of Exclusive" (EOX) or any Status byte, except Real-Time, will terminate a System Exclusive message, and should always be sent immediately at its conclusion.

Format: F0H + 0iiiiiii + data + F7H, where *F0H* is the status byte, and must be followed by manufacturer's identification number; *0iiiiiii* the manufacturer's ID#, 0-127 (current ID numbers are Sequential Circuits 01H, Big Briar 02H, Octave/Plateau 03H, Moog Music 04H, Passport Designs 05H, Lexicon 06H, Oberheim 10H, Bon Tempi 20H, S.I.E.L. 21H, Kawai 40H, Roland 41H, Korg 42H, and Yamaha 43H); *data* any number of bytes (MSB must be set to zero, otherwise it will signal a new status byte; data can range 0-127); and *F7H* an End-of-Block code that terminates System Exclusive status. In no case should other data or status codes be interleaved with System Exclusive data, regardless of whether or not the ID code is recognized.

System Real Time: These codes control the entire system in real-time, and are used for synchronizing sequencers and rhythm units. To maintain timing precision, SRT codes can be sent between any System Common or Channel data sets that consist of two or more bytes. SRT statuses are intended for all channels, and recognized by all units using the interface. If the functions specified are not implemented, they are simply ignored: *F8H Timing-Clock,* and is sent while the transmitter is in Play mode, the system being synchronized with this clock sent at a rate of 24 clocks per quarter note; *F9H* is *Undefined; FAH Start-From-1st-Measure* is sent immediately when the Play button on the master (e.g. sequencer or rhythm unit) is hit; *FBH Continue* is sent when the Continue button (on the master) is hit — a sequence will restart from the point where the sequence stopped on the last Timing-Clock; and *FCH Stop,* a byte that is immediately sent when the stop switch is hit on the master sequencer. The sequence is stopped. *FEH Active Sensing* is used optionally for both receivers and transmitters. It is a "dummy" status byte that is sent every 300 milliseconds (max) when there is no other MIDI activity. If the receiver is expecting the FEH command, it can shut off voices until it receives an FEH or other data.

System Common information is intended for all channels in a system: *F1H Undefined; F2H Song Position Pointer* is made up of three bytes, F2H + 0lllllll + 0hhhhhhh, where l and h represent low and high bytes of a number. The Pointer is an internal register that holds the number of MIDI beats (1 beat=6 MIDI clock counts) since the start of the song. It will start at 0, and then increment once every six MIDI clock cycles until stop is hit. It can be set to any arbitrary number by setting the Song Position Pointer. *F3H Song Select,* comprising two, 8-bit bytes, F3H + 0sssssss, the data byte coding the 7-bit song number; *F4H Undefined; F5H Undefined;* and *F6H Tune Request,* which initiates synthesizer tune routines.

System Reset initializes the entire system to the condition of just having power switched on: *FFH System Reset.* This code should be used sparingly, preferably under manual command only. (In particular, it should not be sent automatically on power up, since it could cause two units connected together to endlessly reset each other.)

It should be noted that the information provided above was issued on August 5, 1983, and represents the current MIDI specification. Further enhancements may be included at a future date — *Ed.*   ☐☐☐

MIDI DATA EQUIVALENT OF MUSICAL SEQUENCE SHOWN RIGHT

Binary	Hex	Comments
10010000	90H	;Note On Event in the first channel (0)
00111100	3CH	;Pitch of middle 'c' (60)
01000000	40H	;Velocity of 64 (mf)—(Note On)
01000000	40H	;Pitch of 'e' above middle 'c' (64)
01000000	40H	;Velocity also 64 (mf)
		;——(time lag to next event)
01000101	45H	;Pitch of 'a' (69) (will carry to next note)
00010110	14H	;Velocity of 20 (p)
		;——(time lag to next event)
01000011	43H	;Pitch of 'g' (67)
00100000	20H	;Velocity of 32
		;——(short time lag to next event)
01000101	45H	;Pitch of 'a' (69)
00000000	00H	;Velocity of 0 (Note Off)
		;——(time lag to next event)
01000011	43H	;Pitch of 'g' (67)
00000000	00H	;Velocity of 0 (Note Off)
		;——(time lag to next event)
01001000	48H	;Pitch of 'c' above middle 'c' (72)
00110000	30H	;Velocity of 48 (a little louder)
		;——(short time lag)
01001000	48H	;Pitch of 'c' (72)
00000000	00H	;Velocity of 0 (Note Off)
		;—— (time lag to next event)
00111100	3CH	;Pitch of middle 'c' (60)
00000000	00H	;Velocity 0 (Note Off)
01000000	40H	;Pitch of 'e' above middle 'c' (64)
00000000	00H	;Velocity 0 (End of first chord)
		;——(time lag to next event)
11000000	C0H	;Program Change in first channel (0)
00000010	02H	;Call Patch #2
		;——(time lag to next event)
01001010	4AH	;Pitch of high 'd' (74)
01010110	56H	;Velocity of 86 (f)
		;——(time lag to next event)
01001010	4AH	;Pitch of high 'd' (74)
00000000	00H	;Velocity 0 (Note Off)
		;——(time lag to next event)
01001100	4CH	;Pitch of high 'e' (76)
01010111	57H	;Velocity of 80 (f)
		;——(time lag to next event)
01001100	4CH	;Pitch of high 'e' (76)
00000000	00H	;Velocity of 0 (Note Off)

STOP PRESS: ROLAND INTRODUCES FIRST MICROCOMPUTER TO MIDI INTERFACE FOR APPLE II AND IBM PC

The new MPU-401 MIDI Processing Unit will be made available with all communications protocol necessary to develop computer/music software. It is hoped that the dissemination of such information will aid in the independent development of new software programs and new computer/music applications.

Currently available for the MIDI Processing Unit are two software programs, one for the Apple II and one for the IBM PC. Each of these programs enable up to eight different musical instruments to be controlled for performance (each one playing a completely different multivoice musical part) by one computer. Not only does the MPU-401 Interface allow such multiple instrument connections (up to 16 separate channels) but, because it is intelligent, it also enables the computer to perform completely different functions while the music is playing, such as program save, load, or other functions such as on-screen graphics.

One of the most significant aspects of the Roland DG MIDI Interface is the relatively low price ($175), and modularity of the system. By utilizing existing microcomputers, and simultaneously using an interface found on many reasonably-priced musical instruments, the new instruments and software applications can be added at will by the user.

For more information, contact: Roland DG Corporation, 7200 Dominion Circle, Los Angeles, CA 90040. (213) 685-5141.   ☐☐☐
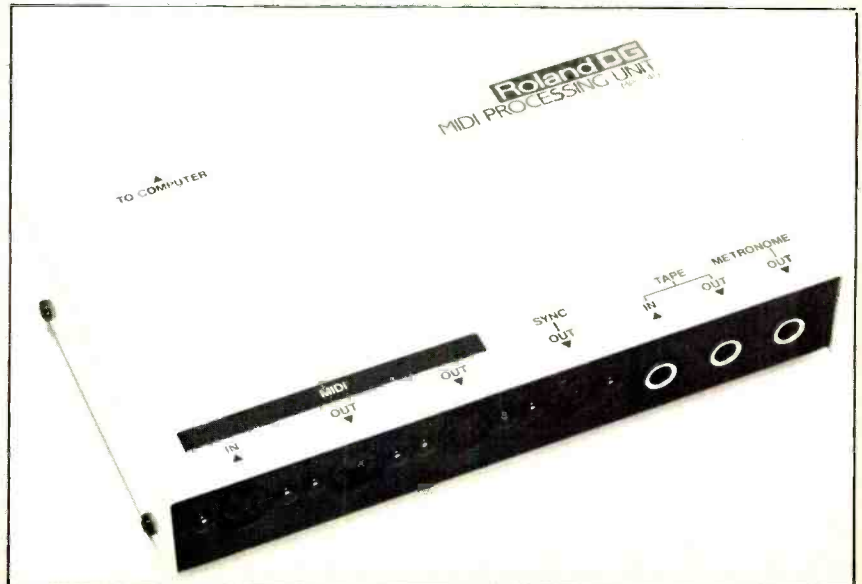
As more and more sophisticated home television and video equipment appears on the market, the quality of television sound has come under increased scrutiny by professionals and consumers alike. In the wake of innovations such as component TV systems and stereo television sound, video technicians across the board — from networks to independent stations to post-production facilities — are taking a close look at their operations. While some are seeking new equipment to improve their audio capabilities, others are looking into new ways of using existing hardware, developing techniques to meet the latest demands of audio for video.

Television's increased audio potential can be traced back several years to two technological advances, now considered to be industry standards. One was the advent of one-inch videotape. Along with improved frequency response and signal-to-noise ratios, one-inch tape — with its two available audio channels, plus code tracks — brought the capacity of recording stereo sound for television and video productions. Added to this, the ascendancy of satellite and microwave technology over the old Telco 5 kHz landline transmission system has ensured that the higher quality audio achieved with one-inch videotape could be transmitted to home audiences.

IMPROVING THE QUALITY OF AUDIO FOR VIDEO

by Adrian Zarin

PACIFIC VIDEO POST-PRODUCTION, LOS ANGELES/Photo by Michael Boshears

Audio and the Networks

In terms of both new equipment and the development of new techniques, all three networks have been active in the audio field recently. Equipment rosters at the various network production centers indicate a tendency to adopt technology developed originally for the music recording industry — particularly in the area of signal processing. As for mixing consoles to be found in major broadcast production studios, preferences are split between boards designed specifically for television sound and those that have made a name for themselves in the record industry, but which have been modified, of course, for the special requirements of TV audio production.

At **ABC's** Los Angeles facility, preparation for coverage of the 1984 Olympics has involved the acquisition of several new items of audio equipment. This recent upgrade included three Ward Beck consoles for three of the four production studios at the network's East Hollywood complex. Flexibility was one of ABC's main criteria in selecting the consoles. Production audio for video differs from the typical music recording session by in placing more demands on the console in terms of bussing and directing signal flow. The complexities of providing various different monitor (or foldback) mixes, feeding a house PA system for studio audiences, and incorporating taped sources such as prerecorded music and commercials into television productions, necessitate a console that can be readily patched and repatched as required. For ABC, Ward Beck television audio consoles answered this need.

"We bought our first Ward Becks in about 1976," says ABC engineering director Don McCrosky. "Before that, our three large production consoles were ones made by McCurdy. They are still in use and very popular, but they are not an in-line type of console; it was more of a building block type of approach with separate attenuators and separate equalizers not in-line with the console [input channel strip]."

The three new Ward Beck consoles selected by ABC are equipped with 36 Model 460M input modules, which are a minor variation on the company's 460L modules. "The only difference is the

IMPROVING THE QUALITY OF AUDIO FOR VIDEO

take off point for public address, isolated feed, and the take off point for solos and so forth, which, depending on the use of the studio, people prefer in different ways," McCrosky explains.

Each input module has two types of outputs; there are left and right outputs which are assignable to four masters. This assignment is a customization of the original Ward Beck design, according to Roy W. Rising, audio systems engineer/production mixer for ABC. "Those outputs called left and right," he explains, "which come out of the panpot when the panpot is in circuit, would normally — the way Ward Beck wires things up — go to odd and even multitrack outputs. We have instead assigned them to the master channel services, of which there are four."

Each input module also has an assignable direct output, which provides a recordable signal from each input channel, and/or to any of 16 submix channels. The submix channels are used to group multiple inputs into a single channel, which then can be assigned to one of the four masters.

The masters, in turn, feed a combination of left, right and mono outputs from the console. In this way, the left and right outputs can be fed any combination of the four masters. The mono output, however, can be fed any combination of the four masters and/or the left and right outputs. "If we wanted a true L+R = Mono output, we could have that," Rising explains. "Or we can have a sort of replica mono, which has the same inputs going into it as the left and right channels, but is not actually a combination of the left and right channels."

In addition, the console has eight auxiliary sends. Normally, four of these are used as feeds to the studio floor for monitoring/ house PA purposes, and four are used as effects sends. Each of the auxiliary sends, however, can be used for any purpose desired.

Summing up the console design, Rising says, "[the Ward Beck] has a lot of capacity in a small space. The main thrust of it is *not* to have dedicated masters and submasters, which are often wasted because they can only perform one function."

To accommodate more than 36 inputs, Ward Beck 26×16 auxiliary mixers are used. To make the add-on mixers as compact and mobile as possible, McCrosky had them built tipped upward so that they stand vertically. The only design modification this necessitated was substituting rotary pots for sliding faders. "It's easier to turn a knob blind and know how far you've gone than it is to operate a fader blind," Rising offers. "Also gravity tends to pull sliders down when you push them up."

Video production's greatest areas of indebtedness to the music recording industry are probably tape machines and signal processing devices. ABC is no exception here. "There's really very little deliniation between audio recorders and broadcast recorders," says McCrosky. "We've used Ampex recorders for a long time, and we also use the same power amplifiers as the music recording industry."

In terms of special effects devices, prominent pieces at ABC's production studio control rooms include UREI Little Dipper filter sets, Lexicon 224 digital reverbs, EMT plates and AKG spring-type reverberation units. Production staffs have mixed preferences, according to McCrosky, when it comes to reverb units. "Some people are diehards and insist on the EMTs," he says, "But there's always a problem because there's never any place to put them.

DON McCROSKY/ABC

Most people, though, are very happy with the 224s."

Unlike ABC, **CBS** chose to go with Neve recording studio consoles when it upgraded the audio production facilities at its Television City complex in Los Angeles some five years ago. While the network, as a matter of company policy, declined to provide specific details of its audio equipment, CBS engineering spokesman Dwight Morris, based in New York, did indicate that the network currently is in the midst of an audio upgrade in its nationwide facilities.

"We're upgrading our facilities to reflect an increasing demand for audio quality on the part of some of the production people," Morris states. "In our studios, we use Neve consoles, which we're rebuilding here in New York. We have a seven-facility studio in New York, and a four-facility studio in Los Angeles. We've also got five television stations around the country with various things installed there."

As for **NBC**, the network relies heavily on custom-made audio equipment in the major production studios at its Burbank, California, complex. The area houses four large audience participation production studios, and a medium-sized studio for taping situation comedies. Each of these studios employ identical custom-built audio consoles, according to NBC audio engineer Ron Estes.

"All the audio boards here were designed and built by NBC about 15 years ago," Estes explains. "While that was a long time ago, they put a lot of thinking into the boards. I don't believe I've ever seen a television audio board this flexible. There is also a huge custom-made jack panel accompanying each board that has over a thousand holes. You can swap modules, mult signals, jump modules, move modules, or do whatever you need to do with this jack panel. The flexibility is great for TV production, where you have to send six or seven different kinds of feeds to the floor or to the PA system, to the foldback system, people monitoring headphones; things like that. It's quite a mess of things to get around."

Each of the NBC-built consoles at the Burbank complex has 53 simultaneous inputs. These inputs are assignable, through a system of submixers, to 20 vertical faders. Ten of the vertical faders are fed directly from the inputs. Each of the remaining 10 faders has four submix inputs associated with it. Volume level on each submix input is controllable by a rotary pot. The four rotary submix pots for each of the 10 vertical faders are located directly above that fader.

The console has one equalizer per vertical fader, which of course means there are no separate EQ facilities for any of the individual submix channels. Equalization, Estes confesses, is one of the console's weak areas. "At the time the board was built," he says, "it was 'state of the art.' That's all there was for equalization in 1964." The EQ facilities that do exist, however, allow the user to select crossover points. Each equalizer boosts 12 dB and cuts 16 dB at 40 and 100 Hz on the low end, and at 3, 5, 10 and 15 kHz on the high end; there is no equalization between 100 Hz and 3 kHz.

The 20 vertical faders on the console are assignable to six rotary submasters, which, in turn, are assignable to two masters: the Cast and Music masters. The two masters are assignable to the board master, 'which,' according to Estes, "is the final output for everything on the board except for two auxiliary faders, having one input each, and the Nemo fader."

The Nemo fader, which takes its name from an old telephone company term, and stands for "Not Emanating from Main Office" — is for controlling the volume levels of "all sends that originate outside of the studio," Estes explains, "which in this case means film, videotape and remote lines. There are six inputs to the Nemo fader, which can be controlled manually or remotely by the video switcher.

"It comes down like a giant tree," Estes says in summing up the console's

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signal flow. "It's essentially a 53×20×6×2×1 board.

Estes further explains that two of the studios at the Burbank complex are slated to be updated in the near future. Originally scheduled for the first half of 1984, the upgrade has been postponed, he says, "because NBC wants to redo the audio and video at the same time, and they're waiting for a new generation of cameras to come out from a certain manufacturer. We've been looking at consoles with added features like camera switching. There's no one that makes an audio console that will meet our needs perfectly. There are just too many things that we need.

"We would like a minimum of 96 inputs on the next generation of boards. Because of space limitations, there are going to have to be some compromises in terms of submixing. When you do this submixing, you have to take into account how you're going to feed the foldback and PA systems. These systems will have to get some inputs in premixed groups, which may or may not be suitable for their purposes. It's a complex situation."

As home base for NBC's *The Tonight Show*, with its healthy quota of live music from both the house band and guest performers, Studio One — where Estes does much of his work — boasts its share of music recording-type equipment. "There's a Lexicon 224, which I use for reverberation on the cast mikes," he says, "and an AKG BX20 that I use for reverberation on the band." Additional outboard equipment includes an EXR Exciter, UREI LA-3 limiters and Valley People Gain Brain II compressor-limiters. "For microphones," Estes continues, "there are RCA 77 DXs on the brass, Neumann KM88s on the reeds, and Schoeps, AKG and Electro-Voice mikes on the drums. Johnny Carson's mike is an AKG C451, and I'm using Sennheiser mikes for applause."

Audio and Independent Stations

At the level of local independent stations in the Los Angeles area, audio equipment selections tend to reflect those of the networks. While some stations — KTLA for instance — have opted for specifically designed TV audio equipment, such as Ward Beck consoles, others have chosen to modify recording studio hardware for their purposes.

As audio engineer for KTTV, Kerry Boggio works regularly on two shows that involve music: *Star Search* and *Thicke of the Night*. "We're using a custom-built Neve console," says Boggio, "and had to modify it for TV work. We sat down with representatives of Neve and told them what we needed; six months later, we had our console. Neve has since come out with a television

console [51 Series — *Ed*], which we're looking at now to replace a couple of older consoles."

Basically, what KTTV required from its TV production audio console was improved flexibility. "We needed more foldback sends," Boggio explains. "Altogether we have 10 sends on the board. Of course, we also needed submixing capabilities. So we incorporated a 16-track submix console into the 32-channel console. We can put all 32 channels into one sub if we wanted to, but we also have the capability to assign channels to all 16 subs. We can then run that into a 16-track recorder, or we can take a 32-track output and run our machine off the submixer."

At KCOP, another Los Angeles independent TV station, audio engineer Tony Alamia has recently finished modifying three ADM consoles for the station: a 32-channel board for its large production studio (used mainly for tap-

RON ESTES/NBC

ing game shows); a 24-channel board for its news studio; and a 16-channel console for the facility's audio recording booth.

"We modified stock boards," Alamia explains, "and put all the masters and patch bays and switchers for the tape decks in the center of the board. Then we put all of our mike inputs on the left-hand side, and all of our line inputs, which can also be miked inputs through patching, on the right-hand side of the board.

"We have a lot of bussing options. I built a custom patch bay and everything comes through that before going into the board. Every input and output can be split, along with the masters, submasters, and all effects."

Stereo Audio for Television

Perhaps the hottest issue in television audio right now is Stereo TV. The growing availability of component television equipment has combined with the vogue for MTV-inspired music video programming to create greater consumer interest in the idea of stereo sound for TV. The major stumbling block to Stereo TV at this point, however, is the adoption of a standard transmission

format. The FCC, in the wake of the controversy that surrounded its handling of the AM Stereo issue, has yet to approve any of the proposed formats for Stereo TV. Meanwhile, the televison production industry is exploring how Stereo TV will affect its techniques and equipment. Just as opinions differ on how far the FCC is from making a definitive move on Stereo TV, so the level of commitment to investigating stereo recording and production techniques differs from network to network.

At ABC, the prevailing attitude is that Stereo TV is still a long way from becoming a reality. "Stereo television may be forced on us by the TV set manufacturers, but I don't think there are that many broadcasters who are that interested in it," says Don McCrosky. "My question always is: 'How much programming would really be enhanced by Stereo TV?' "

According to McCrosky, stereo television is not likely to have any effect on production techniques anyway. "Stereo treatment is going to come in post-production," he says. "Our production methods are not going to change in the slightest — when it comes to post-production, they'll simply pan the audio left and right."

In terms of current production practices, ABC's Roy W. Rising agrees with McCrosky's position. He forsees, however, a renewed interest in live programming as the television industry looks to bring more innovation and excitement to the medium. If stereo broadcasting were to become the norm, live programming would eliminate the possibility of stereo treatment in post-production.

"Post-production won't help us do stereo in a live situation," Rising states. "Producers now often have no concept of 'live.' They wonder why our production consoles are so sophisticated when the most extensive audio treatment takes place in post-production. But the day may come when all this sophistication will have to happen live on the air."

Whatever the future may bring, ABC's current production and transmission facilities are fully stereo-capable, according to McCrosky and Rising. "We could do stereo tomorrow if we wanted to. If there's a standard that somebody would care to adopt," McCrosky concludes, "they could send stereo audio to the transmitters and to the affiliates with no problems."

At CBS, experiments with stereo television began several years ago. "There are no routine stereo productions yet," says Dwight Morris, "but we have done a number of experimental productions to familiarize ourselves with the requirements of stereo for television."

Declining once again to furnish details, Morris states: "Our experiments have included recording at sporting events, and one experiment in stereo music programming — a live stereo recording of the Kennedy Center Trib-

Custom-built 53-input house mixing console in NBC's Studio A provides a wide variety of subgrouping and output options.

ute in Washington D.C. some time ago. We've also done some work at our Technology Center to see what kinds of techniques could be developed to perhaps simulate stereo for programs that were produced monaurally, or where stereo production is either not practical or presently too costly."

Like McCrosky, Morris sees the main burden of stereo sound for television as lying in post-production. "We have found that there is a great deal of flexibility in post-production for creating stereo," he states. "Panning audio channels left and right, recording multitrack audio in sweetening facilities — these are things that are, in some respects, not well known to televison producers and broadcasters in general. We have found that we still have a lot to learn, but we are waiting until we have a reason to start making stereo for practical application broadcasts."

The only network that appears to be routinely engaged in stereo production for current progrmaming in NBC. The network has been taping *The Tonight Show* in stereo at its Burbank complex since last October. The production process, devised by Ron Estes, makes use of both audio channels of one-inch videotape. Before last October, the second audio channel had gone unused. The network is currently broadcasting a mono version of the show, retaining the stereo soundtrack for demonstration and archival purposes.

Each weekday evening, four original one-inch video tapes, with stereo audio, are made of the show. Two of these (a main and a backup copy), are played back at 8:30 p.m. Pacific Standard Time for reception on the East Coast, where it is 11:30 p.m. Eastern Standard Time. As the show finished taping at 6:30 p.m. PST each evening —just two hours before the first airing —there is not time for post-production audio sweetening.

This means that what goes down on tape, and ultimately out to home audiences, is essentially a live stereo mix.

Estes' technique for deriving stereo from NBC's mono consoles stems from experiments he made in the early Seventies as sound mixer for *Midnight Special*, the network's then-reigning pop

music program. Essentially, the system involves using the board's Cast and Music masters as left and right output channels.

"In the conventional operation of the board," Estes explains, you would have all of your Cast mikes — hand mikes, boom mikes, etc. — coming down to the Cast master, and all of your music inputs coming down to the Music master."

To operate the board in stereo, however, Estes uses three of the board's six submasters to group all of the Cast inputs into a left, center and right configuration; i.e., one submaster for left, another for center, and a third for right. The remaining three submasters are used to group all of the music inputs into a similar left, center and right configuration. All of the left channel information — both dialog and music — is then sent to the left output channel (Cast master in conventional operation), and all right information is sent to the right output channel (Music master in conventional operation). All center information is sent to both output channels.

Since the NBC consoles are monaural, no panpots, Estes' system essentially creates 'hard panning' stereo. He has devised, however, several methods for reducing the "hard left/hard right" sound characteristics inherent in the system. For one, he "flips" the reverb channels, sending the reverb for left-

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channel instruments to the right channel, and vice versa."

"Apart from this," Estes says, "you can derive something like panpots by sending a signal to two channels, and playing with the relative volume levels of each channel." Also, on occasions where *The Tonight Show* hosts a large musical group, Estes uses an auxiliary mixer to accommodate the resulting extra inputs to the board. Because the auxiliary mixer — generally a Yamaha 8×2 or 12×2 — is a stereo board, he can make use of the panpots it has.

From the main audio board, the left and right signals are sent into a custom, NBC-built matrix unit, which converts them into two new signals: a sum (L+R) channel and a difference (L–R) channel. The mono sum channel is recorded on track of the videotape. This is the signal that is currently transmitted over the newtork to the affiliates and, in turn, to the viewers' homes.

The difference (L–R) signal is recorded on track #2 of the videotape. This signal, according to Estes, is "anything that is strictly left or strictly right. The center has disappeared because it has been cancelled." The difference signal presently remains in-house for future use. With the advent of Stereo TV, NBC will be able to transmit both the sum and difference channels via their Farrinon audio/video multiplexer, which would convert both audio channels to video subcarrier frequencies for transmission along with the picture. A similar unit at the receiving end would simply demultiplex the video and two audio signals.

The need for a matrix-encoded transmission format, such as the sum and
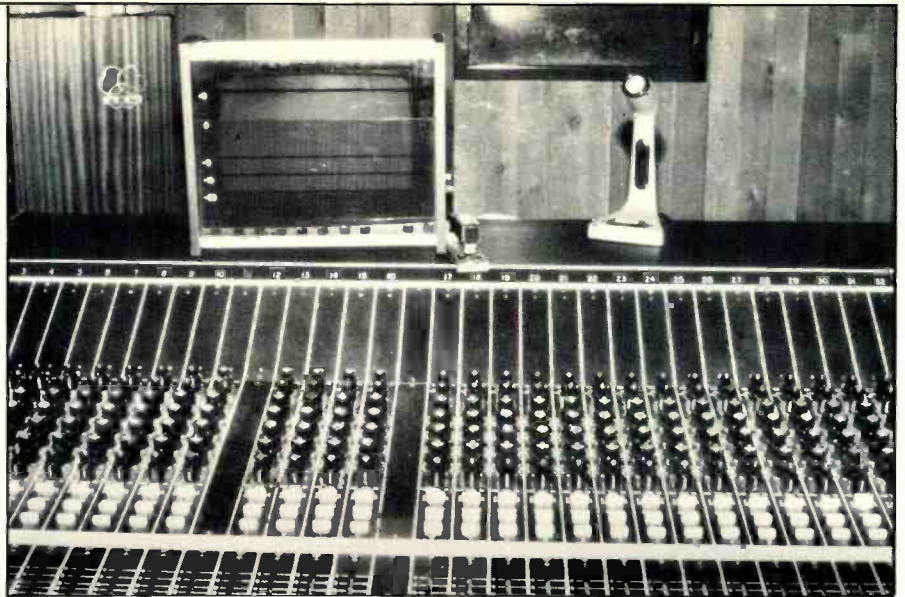
Editel's Audio Sweetening Room houses an ADM console, Ampex ATR-100 two- and ...

... four-tracks, plus extensive outboard processing equipment.

difference code used by NBC, stems from the fact that you cannot simply transmit discrete left and right stereo signals. The transition to stereo television, even it it were to begin tomorrow, would not take place instantaneously. A matrixed transmission system, such as the sum and difference format, provides for the inevitable period of transition from mono to stereo by furnishing a composite L+R signal, containing all audio information recorded, to those facilities not yet equipped to receive stereo audio. Also, in the event of a transmission failure, the chances of being able to get a complete audio signal through to home viewers is increased if at least one of the audio channels contains all the necessary audio program information.

The sum and difference encoding method used by NBC is by no means the only possible format. An alternative technique is what Estes calls the "sum and discrete" technique. This German-developed code places a L+R (sum) signal on track #1 of the videotape, and a right-only (discrete) signal on track #2. Decoding involves subtracting the right-only signal from the sum signal, leaving a left-only signal on channel #1 and a right-only signal on channel #2.

From a transmission standpoint, NBC's multiplexing equipment can handle any encoding format. Currently set up to convert one monaural audio signal to a 5.8 MHz video subcarrier frequency, the addition of a new panel to the Farrinon unit (in a space currently occupied by a blank panel) will enable the network to multiplex a second subcarrier frequency — 6.2 or 6.4 MHz, according to Estes — for transmission.

Audio and the Post-Production Facilities

For a complete picture of the audio-for-video scene, it is necessary to consider video post-production facilities. With a clientele that includes cable and

— AUDIO FOR VIDEO —

commercial producers, as well as broadcasters, the post-production houses and their activities lend a useful perspective to the audio techniques and equipment being adopted by the networks and independents. At Editel/L.A., Compact Video, and EFX Systems—three of Los Angeles' leading post-production facilities — audio sweetening for video has become a vital and growing service, thanks to television producers' increased awareness of the importance of high-quality audio.

"When I first started selling post-production facilities four or five years ago, things were different," reports Eddie Ackerman, vice president/client services at Editel. "I would tell the producers, 'Let's use the sweetening room,' and they would say, 'That's an extra five or six thousand dollar expense; why bother?' Now those same folks are coming back to me and saying, 'How quickly can I get into audio?' So we're in the process of upgrading our audio to match the need there."

Presently, the Editel sweetening room features a customized ADM Technology console, and an Ampex MM-1200 16-track recorder. Mastering is handled by Ampex ATR-102 and ATR-104 stereo and four-track machines. The facility's effects rack includes full Dolby and dbx noise reduction units, a Lexicon Delta T digital delay line, Klark Technik gra-

KELLY KOTERA/COMPACT VIDEO

phic EQ and UREI Little Dipper, along with an AKG reverb system. New equipment acquisitions planned over the next six months include, according to Ackerman, "more Ampex ATRs and an increased sound effects capability; the effects have become a major situation." Longer range expansion plans include an upgrade to 24-track capability in the sweetening room, and the

addition of ADR (Automatic Dialog Replacement), and Foley facilities.

At Compact Video, the situation has been very much the same. "From our point of view," says sales manager Bob Slutske, "we are doing a lot of MTV work and other things where there is more and more *critical* audio throughout. There is just *no* accepting anything less than the best technology can offer right now." Having recently finished expanding its film sound capabilities, the company is now looking to increase its audio-for-video facilities as well, according to Slutske.

The current equipment complement in Compact's video sweetening room centers around an API Model 2488 console and Ampex MM-1200 multitrack tape machines. "In the sound sweetening room for television, we're about halfway between a music recording studio and a film re-recording stage," comments Kelly Kotera, Compact's chief sound engineer for film and videotape. "There are things in here that would be very foreign to a music person — like the Dolby CAT-43 system — and things that music people would be very comfortable with, like the Lexicon Prime Time and Eventide Harmonizer. Having dialog and effects as well as music to deal with, we naturally have equipment and techniques that don't enter into strictly musical work. The whole concept of working with frames

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and feet of film is one such area."

For Kotera, the growing popularity of music programming has been the leading force in the improvement of sound quality for television, which development has led the facility into stereo post-production. "Of course, all of our MTV material goes out in stereo," Kotera states. "Most of those videos are shot on film, and then transferred to video and edited. Our concern, as far as the sound goes, is just retaining the audio quality right down to the end run as much as possible. We make sure the music sounds as much as possible like the original music on the record — not distorted and knocked down [by] several generations of videotape transfer."

Not being tied exclusively to the broadcast business, facilities like Compact, Editel, and EFX already have begun the foray into stereo technology, with the growing home video and cable markets providing the added impetus. "Since videodisks and videocassettes are now being recorded with stereo soundtracks, probably 20% of the shows that go out of here are in stereo," Kotera estimates. "We have been doing stereo for almost a year and a half now because of these applications and things like HBO, which also has stereo capability. Our facilities have two satellite uplinks, which are also stereo capable, and our rooms have always been set up for stereo."

"Music has definitely been a catalyst for intensified post-production in audio," agrees Editel's Eddie Ackerman. In his experience, producers of musical television programming are starting to cover all bets by preparing mono and stereo prints of their shows. Such was the case when Editel provided post-production services on the four

EFX System's video sweetening room features an AMEK 2500 automated console, BTX Soft Touch synchronizer, MCI JH-24 multitrack, and MCI JH-110B video layback.

recent US Festival rock programs that premiered on Showtime cable TV.

"Right now," says Ackerman, "all of the programs that are leaving Editel have a mono audio track for one service that's broadcasting them, as well as a separate set of videotape masters that have the audio separated into channels #1 and #2. So producers are aware that the limitations of their first distribution company are not the *whole* story. There may be a demand for stereo in subsequent distributions, either in the form of videodisks and cassettes, FM simulcast, or actual Stereo TV broadcast."

But the concern with high-quality audio in general, and stereo in particular, is not confined to music programming. Ackerman reports that commercials slated for extensive play on stereo-capable MTV leave Editel with stereo audio tracks, and that even producers of comedy shows are interested in stereo and high-quality audio as a means of imparting a sense of increased realism and audience ambience to their productions.

"There isn't a sitcom delivered to the

networks that doesn't have an extensive amount of audio sweetening," Ackerman states. "This extends to subtle things like laugh tracks and sound effects. The trend now is toward picking up sound effects in post-production, as opposed to recording them on the sound stage. We are looking into digitally processed sound effects now, which is something that has great potential because of the random-access capability of a hard-pack [computer storage] disk to find a particular sound. Plus, we can synthesize the sounds because they're digital — a barking dog can be a Poodle or a German Shepherd."

Editel has developed a method for assembling multitrack soundtracks for sitcoms and other shows that allows maximum flexibility for subsequent re-editing at a later date. Ackerman details the process as follow: "Once the video portion of the show is complete with titles and everything, we take the edited master and lay it onto a separate piece of videotape, retaining SMPTE time-code. Most likely, the copy will be on ¾-inch tape, because of the ability of our equipment to scan it. With the timecode numbers "burned" onto the screen as a visual reference, we record the soundtrack that's available on to our Ampex multitrack. At the same time, we also record the timecode and [59.94 Hz video] sync pulse on channels 16 and 15 respectively; on the remaining channels of the multitrack we drop in sound effects and the the laugh track, each on its own discrete channel. So, if the show goes back to the network and they say we 'over-laughed' the track, we can reconstruct the soundtrack without going through a major expense — we just go back, isolate that one channel, and remix.

"Now that audio is becoming a major thing," Ackerman concludes, "producers are buying that two-inch multitrack audio tape from us. In the past, they would just leave it. With the multitrack tape, they can redo the audio portion of their program at any time, for stereo, quadraphonic, or whatever comes along."

API Model 2488 console, Ampex MM-1200 multitracks, and well-stocked outboard effects racks in the video sweetening room at Compact Video.

EFX Systems in Burbank is a facility that handles both record projects and TV/film post-production. As such, EFX has been in a unique position to observe the growing sophistication of television audio against a backdrop of film and record work. "TV people," says engineer Glenn Berkovitz, "are adopting more of an attitude that's been prevalent in film for a long time, which is that you want to make your audio tracks very clean. You want to keep them separated as best you can to get the optimum results in mixing them. The equalization and overall tone quality of each individual track has become important."

In their video sweetening room, EFX has opted for a music recording console: an automated AMEK 2500. Apart from desiring the flexibility to handle both record and post-production work on the console, EFX felt that the AMEK console was better suited for post-production than many audio boards designed specifically for that purpose. "In terms of signal clarity and directness of signal path," Berkovitz states, "consoles designed specifically for TV are not nearly as clean as any recording console. We also find the equalization on our AMEK is really handy for video. It's got a four-band, fully parametric EQ on each input module, which answers our needs pretty well. We do have the standard outboard notch filters and some additional parametric equalizers, but we can normally do what people need right at the console."

Apart from equipment — which in EFX's sweetening room includes many pieces specifically designed for video post-production, such as the MCI JH-110B one-inch video layback machine, and the BTX Soft Touch synchronizer — Berkovitz finds that many standard techniques from his album work cross over nicely into video post-production. "Harmonizers or delay units like the Lexicon Super Prime Time can do things that really astound TV people," he states. "You can use a delay to do dialog catch-ups for example. Rather than having to lay the misplaced dialog off onto a separate piece of tape, and then re-insert it on the track at the correct spot, you can just use the delay to put the dialog in place. It makes the job quicker and easier.

"To clean up dialog tracks," Berkovitz continues, "we do some things with gating that wouldn't ordinarily be done in a video post-production context — the sort of things you would really use to process drum sounds, say. The situation is similar with equalization. Given our record background, on some occasions we are more aware of what EQ can do to enhance a sound, or remove unpleasant overtones."

Berkovitz is quick to point out, however, that a music recording background may be useful in post-production work, but it's no substitute for a thorough grounding in the highly specialized equipment and techniques video entails. "We've had a couple of video soundtracks come in that were mixed by music people," Berkovitz reports, "and while they weren't *wrong*, they were not quite right either. Video work demands a very different kind of awareness. At EFX we have separate staff people for records and for TV and film work. We've found that it takes quite a bit of adaptation to switch from one to the other.

"As TV people have become more sophisticated," he concludes, "they realize what post-production can do. They are more critical. You really have to concentrate on the sound in video work now. There are no quick fixes for TV as there have been in the past. TV people are more demanding now."

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In both production and post-production, it would seem audio for video has taken great strides in recent years. At the present time, flexibility is an operative concept in television sound as the field faces an encouraging but still-uncertain future. The future of audio for video techniques is clearly tied up with the fate of Stereo TV — a question which depends on such outside variables as the consumer TV hardware market, and the FCC's decision on standards. Meanwhile, in looking to improve television sound quality, video professionals are searching for ways to be prepared for every eventuality. ■■■

AMBISONIC SURROUND-SOUND TECHNOLOGY FOR RECORDING AND BROADCAST

by Nigel Branwell,
Audio + Design Recording, Inc.

Many engineers and producers would consider that the biggest limitation faced by those of us involved with recording and broadcasting conventional high-quality audio is no longer distortion, noise, or poor frequency response. It is rather the "cardboard cut-out" sound images caused by the limitation of having just two loudspeakers available for stereo reproduction. Good as stereo may be, it cannot capture the three-dimensional world of sounds. Imagine birds singing overhead, applause from an audience situated all around, or the sound of footsteps beneath you as you walk; imagine the all-surrounding reverberation of a large concert hall.

This is what the Ambisonic surround-sound system of recording and reproduction can achieve. Ambisonics works by using the psycho-acoustics of human directional hearing to capture all directions of sound around the listener; its principles of operation are based on over a decade of research and development. Unlike "Quadraphonics," which was basically two stereo systems — one placed in front of the listener, and one

behind — Ambisonics can create convincing and accurate localized phantom images even between side loudspeakers. Furthermore, the listener can move around the listening area, and face in any direction without losing the sound image.

Ambisonics is indeed a total systems approach to capturing surround sound information, and is aimed at meeting the widest range of professional and consumer requirements — from the needs of a listener with a pocket AM radio, to the audiophile listener with a custom multispeaker home system. Not only is it compatible with existing mono and stereo media, Ambisonics also offers practically unlimited capability of meeting both near and distant future needs. Beyond stereo, the continuing interest in four-speaker sound can be satisfied by Ambisonics via existing records, tapes, and radio.

The UHJ (standing for "Universal HJ") system of encoding directional sounds into two or more transmission channels, developed in conjunction with the British Broadcasting Corporation, conveys the Ambisonic soundfield

to the final listener, and was designed to achieve outstanding mono and stereo compatibility. No mono listener should lose any important sonic information, no matter where the producer decides to place it in space. In addition, a stereo listener tuned into a UHJ broadcast not only hears a full-width stereo presentation, but also enjoys an enhanced sense of depth and focus not possible with conventional stereo transmission systems. And, of course, the listener equipped with psycho-acoustically optimized Ambisonic surround sound decoders hears the precise pattern of surround sound intended by the producer.

UHJ Two- and Multi-Channel Systems

In its simplest form, UHJ uses the same two recording or transmission channels as conventional stereo, utilizing both amplitude and phase to convey a full 360-degree, horizontal sound stage. Further enhancement of the quality of sound images around the listener can be achieved by adding a third recording or transmission channel to the basic two-channel encoding. The effect of full-sphere portrayal of directionality, including sounds above and below the horizontal soundfield, can be conveyed by the addition of another supplementary channel for height information.

As will readily be appreciated, this heirarchical system provides the option of using two, three or four channels with UHJ, depending on how many channels are available for storage or transmission. For example, in FM broadcasting, a recent decision by the FCC has made available three- and four-channel multiplex FM. Naturally, broadcasters not yet ready to update their equipment can still broadcast UHJ in its two-channel

POTENTIAL PLAYBACK ARRANGEMENTS FOR AMBISONIC-ENCODED MATERIAL

FM/AM TUNER

TURNTABLE

CASSETTE/COMPACT DISC

REEL-TO-REEL

CONTROL UNIT

TWO OR MORE CHANNELS

POWER AMP

UHJ DECODER

POWER AMP

LISTENER

ALTERNATIVE SPEAKER LAYOUT (ALL CONTROLLABLE FROM DECODER)

AMBISONIC PLAYBACK CONFIGURATION

AMBISONIC SURROUND-SOUND

form. While present-day LPs, cassettes, and Compact Discs convey two channels, the CD standard allows for future marketing of disks with up to four audio channels. Already, a number of two-channel UHJ Compact Discs are on the market.

All Ambisonic systems, whether utilizing two, three, or four transmission/-storage channels, can reproduce surround sound through four or more loudspeakers. Most listeners will probably prefer to use just four speakers for the time being; however, even better results are obtained if six or more speakers are used, since the Ambisonic system does not convey sounds for just four loudspeaker channels, but a *total*, all-direction soundfield. For this reason, sound can be tapped off from any direction to re-create the soundfield via practically any number of loudspeakers.

International patent rights covering Ambisonic technology are held by the NRDC (National Research and Development Corporation) in Britain, and incorporate inventions from the USA, Japan, and Britain. All aspects of this technology are available to manufacturers under license.

Studio and Recording Technology

The total soundfield of Ambisonics is "captured" in the studio in what is referred to as B-Format. This consists of four signals: W, X, Y, and Z. W is an omnidirectional pickup of the soundfield consisting of sounds from every direction with equal gain, while X, Y, and Z are bi-directional (figure-of-eight) pickups of the soundfield pointing respectively forward to back, left to right, and up and down. For horizontal surround soundfields, the Z signal is set to zero, only the W, X, and Y signals being required. All four signals represent a total, three-dimensional and direc-

Calrec Mark 4 Soundfield Microphone with cover removed (right), and control unit, capable of capturing a total 360-degree soundfield.
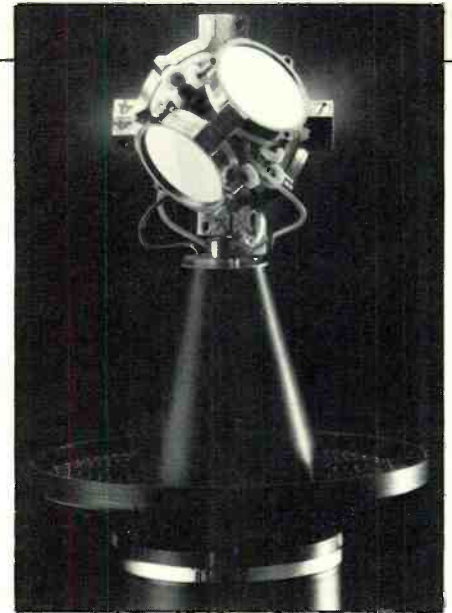
tional soundfield. It should be emphasized, however, that these four B-Forward signals have *nothing* to do with the loudspeaker feeds in a four-speaker monitoring array.

B-Format signals are produced by most Ambisonic studio production equipment, and for dissemination to consumers is converted to UHJ formats two-, three-, or four-channel by means of a UHJ encoder. The studio handling of Ambisonics and its conversion to a usable consumer format for broadcast or commercial release is illustrated in the accompanying diagram, which shows several of the options involved.

So far, Ambisonics technology has led to the development of a Soundfield Microphone manufactured by Calrec Audio, and which has been praised by many users for its high degree of realism and accuracy; a Transcoder that offers both Encode (B Format to two-channel UHJ) and Transcode (four-channel or "Quad" to UHJ two-channel); and a Multi-Track Pan/Rotate unit — the latter two devices being manufactured by Audio + Design. The Pan/Rotate unit enables a whole new range of creative sound manipulation techniques, such as full-range positional control to be achieved for both single (mono-miked) sound

sources, or composite soundfields.

Other Ambisonic units currently available include the Calrec UHJ Encoder, which encodes four-channel B Format to two- or three-channel UHJ for broadcast or disk cutting; the Audio + Design Converter, which enables a mixing console's panpots to be used for soundfield localization — thus freeing up the Pan/Rotate unit for dynamic panning effects; the Minim AD12 Decoder, which decoded B Format or UHJ two-channel material; the Audio + Design Control Room Decoder, which

UHJ IS A HIERARCHICAL ENCODING/DECODING SYSTEM BASED ON AMBISONICS SURROUND-SOUND TECHNOLOGY

AMBISONIC SURROUND-SOUND

will decode two- and three-channel UHJ signals, and allows comparison with original four-channel B-Format signals; plus several UHJ decoders designed for consumer use.

Although the Calrec Soundfield Microphone has been available for some time now, October of this year saw the introduction of a Mark 4 version, whose availability also coincides with release of the above-mentioned Ambisonic encoders, decoders, and effects units.

As many readers may already be aware, the Soundfield Microphone system comprises a microphone body and electronics, plus associated control unit. The body houses four mike capsules arranged in a tetrahedral array, with electronic compensation to remove the effects of capsule spacing. The Soundfield mike is designed to capture accurately the sound that exists at a point in space; in other words, it comprises a three-dimensional coincident "pair." Of primary importance is the system's ability to capture a *complete* spherical soundfield that can be manipulated by panning and steering the signals in both the horizontal and vertical planes. This manipulation can be accomplished in either real time — as perhaps the ultimate boom mike — or later in post-production from the B-Format signals recorded on tape.

Azimuth and Elevation controls govern rotation and tilting of the microphone's orientation in space, while a Dominance or Zoom control allows the direction of dominant sound to be moved to the front or back. (For example, to give an orchestra or band dominance over audience noise, and enable not only manipulation of the effective polar pattern — from omni through all the intermediate cardioids to figure-of-eight — but also the amplitude of the respective dominant signals.) Both B-Format inputs and outputs are available on the control unit, as well as conventional stereo outputs.

It is important to realize in the case of the Soundfield Microphone, that quite apart from surround-sound applications, for conventional mono/stereo reproduction the microphone's polar diagram may be synthesized either in real time, or subsequent post-production from B Format.

Ambisonic production has perhaps been held back because, until now, only the Soundfield Microphone has been available for recording live music, drama, speech, etc. Since most of the recording and broadcast production industry relies on mass-market multi-track derived music, Ambisonics has not had enough to offer. However, the recent introduction of processing equipment, including the Multi-track Pan/Rotate, Converter, and Transcoder units mentioned above, should change this situation, since they enable mono-miked sources to be manipulated within an Ambisonic soundfield.

The Pan/Rotate unit features eight continuously rotatable sine/cosine panpots, plus a switchable control that will rotate the entire soundfield through 360 degrees. The panpots can be switched pre or post (before/after) the rotate control. Apart from the eight inputs from the mixing console, there are two external B-Format inputs — one patched before and another after the rotate control — which may be used to cascade these or similar units. Each panpot includes a "radius vector" control that allows sounds to be panned through the listener's position from one side of the soundfield to the other — quite apart from the ability to move sounds around the circumference of the soundfield.

The B-Format output from the Pan/Rotate Unit can be sent to a four-track recorder, or direct to a suitable encoder to produce a UHJ two-channel master tape for broadcast or disk cutting, or to a three-channel encoder for enhanced surround-sound broadcast.

The Converter unit allows a conventional console's panpots to be utilized for soundfield localization. The unit contains two independent B-Format groups, each with separate outputs. Five inputs feed each coverter group, and are designed to be connected to four console output groups, plus an echo send bus. The echo send supplies a mono reference signal, while the group inputs provide level information that is "converted" with the mono signal into a B-Format signal. Thus, a console panpot can be used as an Ambisonic localization control, simply by turning up the echo send level, and routing the mono source to two of the four groups.

The Transcoder takes four inputs (corresponding to a front and a rear stereo pair — for example, an existing "quad" mix) and "transcodes" these into a UHJ two-channel signal. Control of the width of front and rear sound-stages is possible, and may be varied
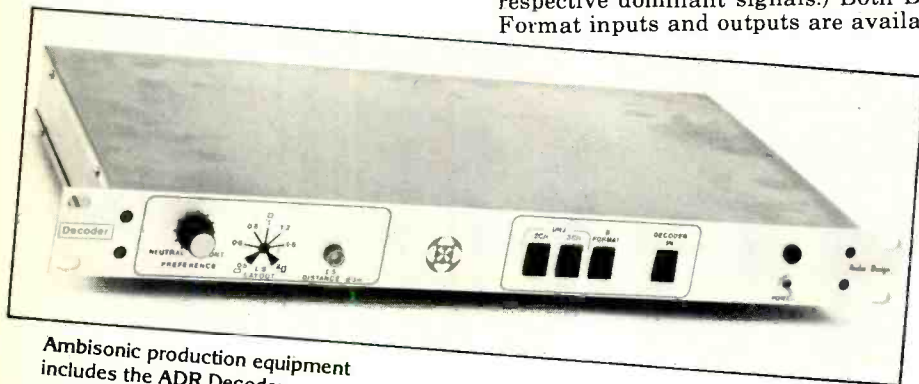
Ambisonic production equipment includes the ADR Decoder . . .

. . . UHJ Transcoder/Encoder
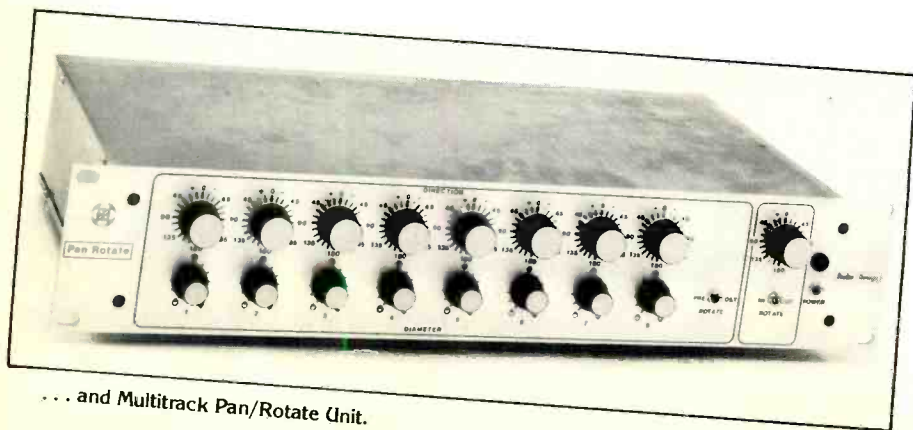
. . . and Multitrack Pan/Rotate Unit.

between 0 and 180 degrees in the front sector, and 0 to 150 degrees in the rear. Panning between the front two groups via the console panpots moves the signal across the entire width of the front stage.

Enhanced Creativity

Using the equipment described above, the creative effects possibilities in Ambisonics, including non-realistic events such as sound sources moving around and through the listener's head, are practically endless. There are few pitfalls, however. It is important to remember when mixing Ambisonic material which may be monitored by the consumer in stereo that although the system is *exceptionally* mono/stereo compatible, signals panned around the rear sector of the soundfield, for example, will be "collapsed" to the front in stereo. As a result, care should be taken with placement to avoid cluttering the stereo image. This can easily be achieved by simply monitoring in stereo from time to time as a check of surround-sound/stereo compatibility.

Because Ambisonics uses phase and amplitude information to localize a source, rather than just amplitude, the image is far less dependent on the listener's position (in both stereo or surround sound monitoring layouts), which means that the listener can move around the room without losing image stability. In both stereo and surround-sound monitoring arrays, the resultant image is more stable and can thus be localized better. Furthermore, the apparent stereo width of an Ambisonic recording played back in stereo is perceived to be *wider* than the speaker separation; in fact it is possible to produce stereo effects that appear to move round the room without the benefit of an Ambisonic decoder. (Similar results can be obtained using stereo headphones.)

Monitoring setups for horizontal surround-sound (that is, without height or Z-channel information) is quite simple, a suitable decoder plus four speakers and additional amps being all that is required. (Six or more speakers are necessary for full sphere or "periphonic" reproduction.)

It must be stressed that Ambisonics has the potential of constituting at least as an important step forward for audio as digital technology. In its various hierarchical, intercompatible forms, Ambisonics material may be stored digitally or by conventional analog means, and broadcast techniques using phase-quadrature modulation of the third channel have been successfully demonstrated in Britain by the Independent Broadcasting Authority, and is now permitted in the United States.

All the hardware to record and disseminate Ambisonic surround-sound information is now in place, with more on the way. It is now just a matter of time before Ambisonics will be hard regularly on record, on the air and, perhaps, with video. ∎∎∎

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The Warehouse 24-track Production Facility

THE CREATIVE ART OF JINGLE AND COMMERCIALS PRODUCTION

New York City's Lucas/McFaul Working on Multitrack Music Sessions for AT&T, GE and Kentucky Fried Chicken

by Robert Carr

"I'll do jingle dates for a while to pay the rent." How many times have we heard musicians say that? For a long time, "serious" players, producers, and engineers have tended to look down on a livelihood derived from creating commercials. Yet, ironically, it's a fact that these days commercials, jingles sessions, and industrial presentations comprise most of the work passing through a majority of recording studios. In many cases, the writer of the recorded music may also be the producer or the engineer on the date. And such people who continue to make a living in this way realize the high level of expertise required to gain and maintain a modicum of success as a music supplier.

One company that has managed to remain at the top of the commercials profession is New York City-based Lucas/McFaul, writers of such memorable pieces as AT&T's "Reach Out, Reach Out and Touch Someone," Kentucky Fried Chicken's "We Do Chicken Right," General Electric's "We Bring Good Things to Life," and scores of

major commercials for products such as Dannon Yogurt, Special K, Mountain Dew, Right Guard, and Soft 'N' Dri.

Not only do David Lucas and Tom McFaul love their jobs and the individuals they work with, there is a good chance that more people get to hear their 30- and 60-second masterpieces in one day than most hit records achieve in a month or more. As a result, Lucas and McFaul have built a small recording empire on Manhattan's West 46th Street, which boasts offices for writing, business, sales and maintenance staffs, an eight-track demo studio, plus a full-blown 24-track recording facility for personal (and occasionally for their friendly competitors') use.

Genesis of a Commercial

From such a successful vantage point, David Lucas and Tom McFaul are eminently qualified to discuss the mechanics and creative considerations of jingle writing and recording, from agency conception to airing on radio and television.

A jingle starts with a client, its product, and an advertising agency hired to develop a campaign. The agency, in close association with the manufacturing clients, spends months learning about the product, conducting market research, developing a campaign strategy that's acceptable to the myriad of professionals who have to lend their final approval and, when necessary, supervising the scripting and shooting of the commercial's visual portions. Once all this pre-production has been completed, the agency turns its attention toward the jingle house or, more accurately, several jingle houses, to see who can translate the ideas and concepts into a miniature musical extravaganza that always "has to be done yesterday!"

Regardless of how good a jingle writer may be, or how established the reputation of the supply house, most commercials projects are awarded only as a result of competition. An agency developing a new campaign usually goes first to the supplier that produced the last

successful arrangement for that client, and asks them, along with two or three other houses, to submit their best musical ideas. In the case of Lucas and McFaul's General Electric campaign, where the company is writing new songs that require the GE tag to be included on the back, the agency does not solicit different writers for each new version.

"Even if they don't like the first thing that we write," says Tom McFaul, "they won't run off in the middle of the GE campaign and go some place else for the new part that has to be composed. There is a loyalty on both sides to a certain degree."

According to McFaul, successful writers win about one of every 10 jobs for which they compete. Although the advertising agencies subsidize the costs of a competition — expenses for coming up with roughs of the jingle and producing a demo — he feels that the process still takes its toll. "We generally expect competition, but not unending competition. The ones that go on forever cost us money. We're basically getting out-of-pocket expenses, but we're not making any profit. The situation has gotten better in recent years; it used to be 15 people writing and submitting demos for one project."

Lucas and McFaul generally turn down a job if there's more than three competitors. "Some people would think that a creative fee is a lot of money," adds David Lucas, "because the numbers look good on paper. But when you're winning one project in every 10,

that one is paying for all the effort and overhead that it took to get the other nine finished. A lot of money is wasted."

The client's many representatives and the hierarchy of agency involved in passing judgement on the music piece generate an additional burden to a project. "The process can be frustrating sometimes," Lucas notes. "The trick is to work for the boss, or whoever is temporarily the boss, or else you're constantly getting the run-around on approvals."

Watching the Clock

To compound matters, the time frames within which jingles need to be written are short. A typical example runs something like this: The agency may present lyrics to the supplier on Monday; expect to hear songs on Wednesday; and listen to the full-blown recording of rhythm section, finished vocals, and orchestration on Thursday.

Lucas points out that the pressure is passed down to all the suppliers — in the case of a TV commercial, to the video or film director, editors, etc. — not to just the music people. Because an agency creative group usually services more than one account at a time, it must work out strict schedules to distribute the work load. "They may allot only a week for a project, and then they're on to the next one. The business moves very rapidly," he concedes.

Would scheduling more time for music writing and production improve the quality of jingles? Lucas and McFaul feel it would, to the extent that there would be less repeating of a session. "The advertising business has an old saying," McFaul notes. " 'There's never enough time to do something right the first time, but there is *always* enough time to re-do it.' There's an awful lot of 're-doing' it."

Yet as far as the actual composing is concerned, pressure sometimes makes for better music. Very often, the first intuitive response proves to be the best, McFaul offers. "I may work laboriously on a 30-second spot for hours, which will come out good, but it will usually be a more complicated spot than the ones that come quickly or intuitively. Generally speaking, what I come up with first is the best, and most immediately apparent."

"You can't doubt yourself," Lucas emphasizes. "Just because something comes quickly, the tendency is to think that it *can't* be that good. But you have to accept the fact that what is easy for you, may not necessarily be easy for anyone else."

The most important asset associated with extra time is the luxury of being able to walk away from what has been written, and then return for the second listen a few hours later. "Most of the time you can immediately hear what is wrong with the original piece," McFaul continues. "But when you're working quickly under pressure, dumb little things slip by that could be corrected

easily by someone else hearing it. We have to rely on the feedback of each other or another peer to get their clear, objective reaction before the tune is presented."

Anyone who's prepared a commercial knows the feeling of hurrying through a session only to realize the next morning that the tracks don't work. Conversely, there's always the joy of discovering some truly inspired performances, writing, and engineering that essentially went matter of factly all day long in the rushed atmosphere of the recording session.

"Hurrying can get you to do something miserable or something great," says Lucas. "Whereas if you had more time, you wouldn't get those peaks and valleys. You might never write anything as brilliant as some of the pieces you crank out under pressure but, in the long run, you'll consistently write better material."

McFaul tempers that thought with the statement that too much time would work the opposite way: "There should be no reason to make a big project out of writing a jingle. They're little songs that go by fast, feel right, and have a little hook that gets the listener right away."

And they're working faster all the time. Thirty-second spots are rapidly replacing the 60-second commercial as the dominant advertising length, which makes a normal song form unsuitable, and limits the possibilities available to

"Club singer," "drummer," "composer," "record producer," and "record hawker" all describe steps in the musical evolution of **David Lucas**. But none seemed to offer him the fulfillment he was looking for. In 1963 he joined his cousin's firm, Don Elliot Productions, to learn the jingle business. By 1966, Lucas felt sufficiently confident to start his own New York-based jingle company, David Lucas Associates. Since then he has taken time from his hectic writing schedule to produce albums for the Alessi Brothers, and three Blue Oyster Cult albums, with *Agents of Fortune* reaching Platinum status. As a prolific songwriter, Lucas finds jingles to be an appropriate career.

While attending the University of Illinois in the early Sixties, **Tom McFaul** focused on the study of ethnomusicology, Baroque, and Romantic music. Upon graduation, he chose to express himself through various club bands, music copying, and song writing, because he was not aiming for a career in composition or arranging. Yet while playing in Buffalo, New York, with an avant-garde rock group called Think Dog, he met David Lucas, and they both discovered many similarities in taste and writing styles between them. This friendship brought McFaul to New York City, where he joined David Lucas Associates in 1969 as a writer and arranger and, in 1973, as a partner to form Lucas/McFaul, Inc.

CROWN PZM

SOUND LOCK

PILLAR

GUITAR

● BEYER M88

GUITAR ● M88

● M88

OVERHANG WITH TRAP

FENDER RHODES

DI

CARPET AREA

EASTLAKE CONTROL ROOM

C414EB

VOCALS

C414EB

STONE SURFACE

TRIPLE-GLASS WINDOW

STUDIO LAYOUT FOR "KENTUCKY FRIED CHICKEN" RHYTHM TRACKS

CONCRETE FLOOR

DRUM MIKING:
O/H: C452EBs
TOMS: SENNHEISER MD421
SNARE: MD421
BASS: AKG D12E

BASS DI

MIRROR SURFACES WITH DRAPES

structure an effective emotional build. Many agencies are tending toward the big song — a commercial composed of words and music with no announcer reading copy — as the primary vehicle for reaching their respective markets. A 60-second time frame provided plenty of opportunity to write a little song: a verse, maybe two, a chorus, a little bridge, and an out-chorus.

"Unfortunately, people are still expecting 60-second structure in 30 seconds," McFaul laments. "In 30 seconds you cannot start out easy, develop the theme, throw in a surprise, deliver the chorus, deliver it a second time, and still compete with a regular three-minute radio song. Instead of telling a story, the 30-second format only makes an *impression*; you get how the music feels, and maybe some repetition of the hook, [and] some kind of memorability out of it."

Further restrictions often come in the form of agency conceived and scripted lyrics. A lyric that the agency thinks is terrific can generate friction if Lucas and McFaul feel that the words are not conducive to the appropriate piece of music to be written. Lucas explains: "Just by scanning the lyrics, and the market research, you know what the jingle is *supposed* to sound and feel like, and whether the lyrics and music match well. To make some lyrics work, you may have to readjust them. If the agency doesn't allow that bend, because they've committed themselves to the original version, you can get stuck. It's

frustrating.

"We prefer to work with agency people who are willing to change their view point when that's best for the project," he continues. "An agency that considers a supply house as people who should just do as they're told is not going to get the best for themselves. The experienced suppliers can offer valuable input, too. On the other hand, the music is only a *vehicle* to get that product sold. The writer can't lock himself in an ivory tower and 'groove' on the music. The account executive knows important aspects about the campaign's objectives and, therefore, the music people *have* to listen to him. The only teams that work well — at least where we've had the best results — are the ones that have input going in all directions."

Agency/Studio Cooperation

To help bridge the communication gap between these two creative factions, the agency must supply the jingle house with all the information necessary to write an effective tune. This information comes in the form of a product sample, and a fact sheet that tells what the product is; what it does; and who is going to buy it.

"A campaign must take into consideration the competition and the consumer," says Lucas. "I imagine the appropriate consumer for that product in the place where that product is sold, and try to figure out what will make them reach for that product instead of another. In the case of Kentucky Fried

Chicken, for example, I would imagine myself driving down the road and seeing their restaurant. I imagine myself as the consumer, and focus on what would make me pull into a Kentucky Fried Chicken [restaurant]. The product is there for you as much as for anybody else."

McFaul, writer of the KFC piece, never got to record it in the rock-and-roll format he originally envisioned. "That's been a little disappointing to me," he concedes, "but fortunately the song proved to be flexible enough to get in and out of the different styles we've had to do, although it hasn't been as easy to work with as 'Reach Out,' or the Pepsi song, or GE."

To date, AT&T's "Reach Out" commercial has been redone 150 ways, and each version has such a unique beginning that the viewer usually doesn't recognize them until the tag comes up. The ad campaign has been able to last for five years, and is now well into its sixth year. The secret is versatility, Lucas and McFaul consider.

The optimum jingle is one that can "bend" in many different ways. An advertising agency and the client commission arrangements in various styles, especially when it is unfolding a massive campaign for a major product that's projected to run for a while. Marketing studies reveal what members of the potential audience a client wants to reach with the advertising.

Lucas and McFaul prefer writing for a specific audience, because the guide-

STUDIO LAYOUT FOR "KENTUCKY FRIED CHICKEN" HORN OVERDUBS

C414 — BARITONE SAXOPHONE
C414 — TENOR SAX/FLUTE
C414 — ALTO SAX/FLUTE
U87 — FRENCH HORNS
U87
U47 — TRUMPETS
U87
U47
TENOR TROMBONE
BASS TROMBONE
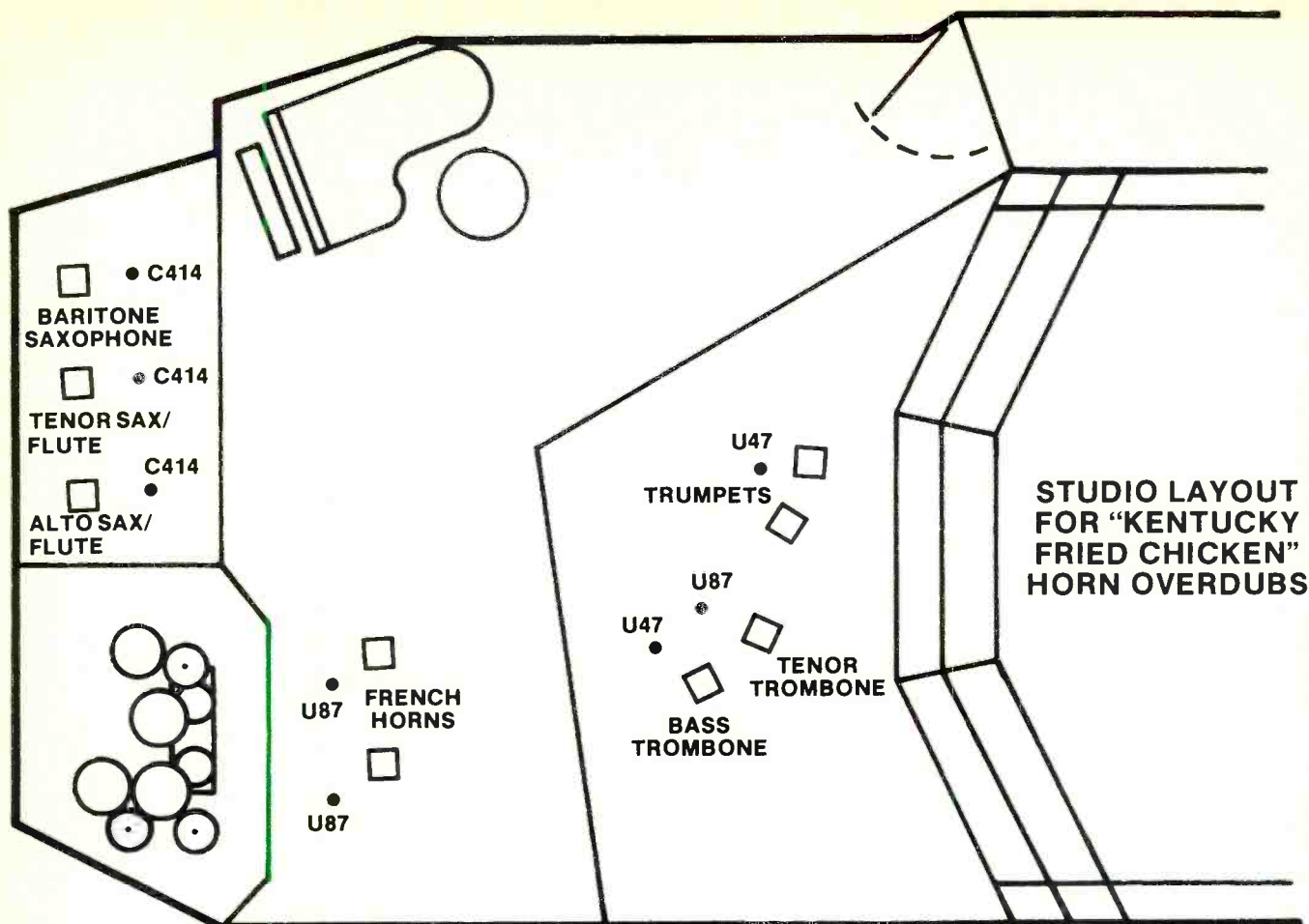
lines are more defined. A period piece should always be perfectly Fifties, perfectly baroque, perfectly Jeanette McDonald/Nelson Eddy, or whatever the setting demands. But the flagship song for a big advertiser has to be in the mainstream of contemporary pop music, simply because contemporary, middle-of-the-road music appeals to the most people. However, an AM-radio type of song holds the danger of drawing the focus away from the commercial, which defeats the original intent. A song has to enhance the audience's hearing of what's being said, and the seeing of what's being shown, without getting in the way. In fact, if the music appears invisible, that's alright, too. "Music is emotional support that adds feeling to the commercial," says McFaul. "A commercial is similar to movie scoring, in the sense that the music helps move the plot from one point to another without drawing attention to itself."

A jingle written in a very particular style may contain a certain harmonic progression or intervals that can't be changed, because those parts provide the hook. Yet those intervals that characterize the song's original style to such a high degree may not be appropriate in the new rendition of a tune. Looking at Kentucky Fried Chicken again as an example, the hook part of the song contains a major-seventh interval that would never be heard in a country song. "If I had known in the beginning that we were going to do so many country

versions," says McFaul, "I probably wouldn't have written that interval into the song. It proves to be a difficult thing to work with now."

Working to Demo Tapes

A demo copy of a newly conceived jingle is submitted to the agency on cassette. Lucas and McFaul record their own eight-track piano/vocal demos inhouse. Usually the composer of the song lays down a minimal-accompaniment part on a keyboard instrument or guitar, and doubles the voice a couple of times. Some new, enthusiastic production houses spend great amounts of money on demo tapes, and often that extra effort helps its chances. But Lucas emphasizes that the writer has to consider who at the agency does the listening. "The consumate pros on Madison Avenue can't be fooled by embellishments. They're looking at the song, and most of the time they'll pick the same ones that we know are the best."

To maintain a high level of consistency and quality in their finished product, Lucas and McFaul also built a 24-track studio into their office complex, known as The Warehouse. Jokingly, Lucas says he wanted to build the studio because he's too lazy to go out of the house. However, he quickly shares some of his more serious reasons: "I don't like hunting all over town for an open studio, or getting bumped if I run over [the allotted time]. I don't like having different engineers for every session, or paying someone else $30,000 a month for

equipment that they'll own, and employees that they choose. We're paying for equipment that we own; supporting people that we like. And you can't beat the efficiency and consistency.

"Clients and agency people know exactly what to expect when they get here; they know where the coffee is, and they know the people who work with us. It's the little conveniences that make the difference."

But when it comes to musicians, Lucas and McFaul don't have a staff of regular players as many other production houses strive to establish. McFaul explains: "The excitment of this business comes from the variety of players that we hire. The combination is always different, and always great. You'll find the same people on our commercials, for example, that play Steeley Dan dates."

The arrangers write out the music for the session following a general lead sheet/rhythm chart format, because there are never any head or improvised sessions. McFaul points out that "if you don't give high-quality professional musicians anything definite to play, they tend to get confused. Many people wonder, 'How can I write a guitar part for a great player like David Spinoza?' Yet that's what he wants. The more you write, the *better* the music gets. If you write something dumb, you can always throw it away or change it."

"But actually," Lucas adds, "we're hiring top musicians to make any song sound great. If a silly song is appropriate for the commercial, then you hire

HARP

C414

C414EB

CELLI

C414EB

VIOLAS • C452EB

VIOLAS • C452EB

• U87

U87

VIOLINS

U87

• U87

STUDIO LAYOUT FOR "KENTUCKY FRIED CHICKEN" STRING OVERDUBS
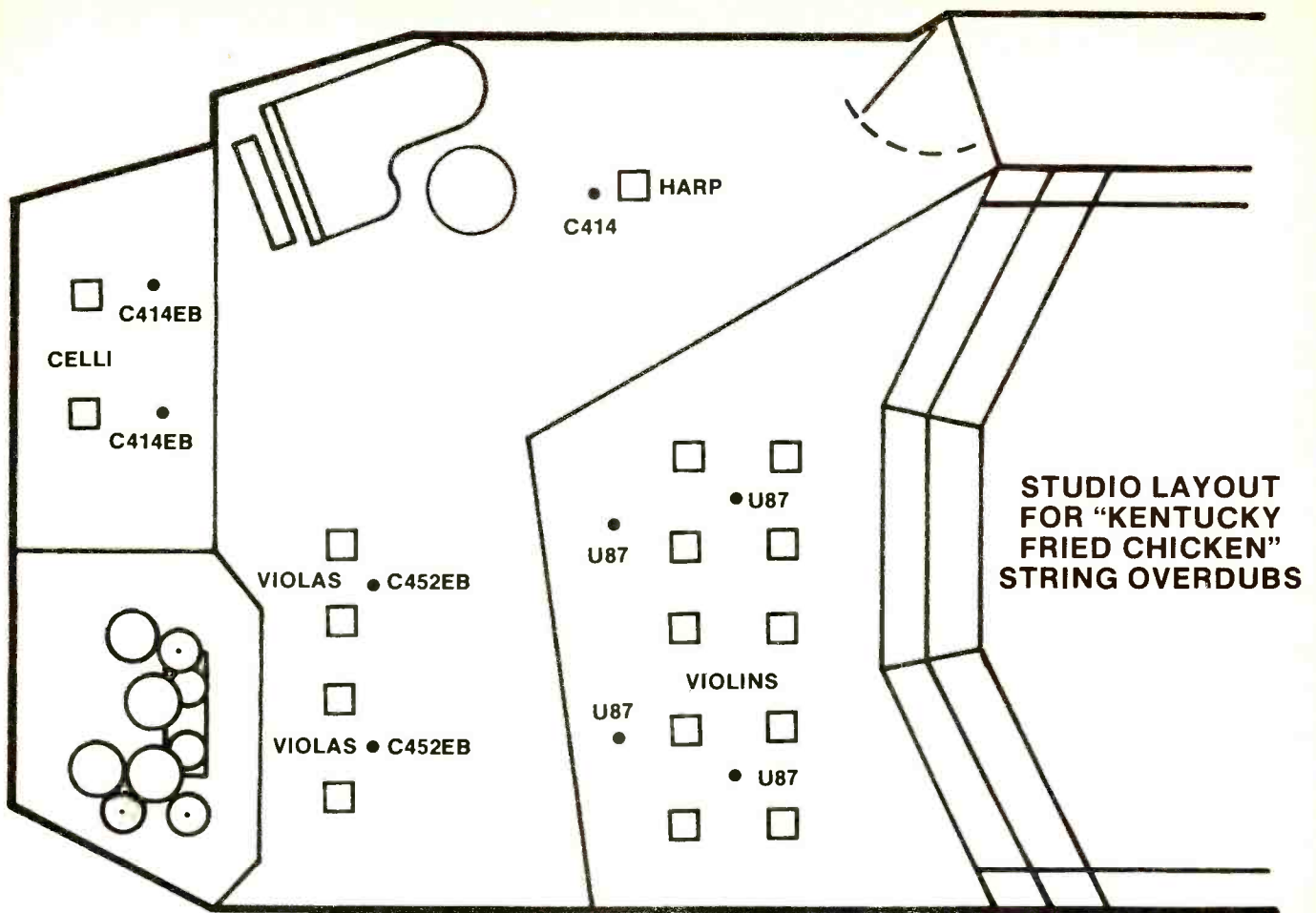
the best to make it sound great. Music that's played well and feels good never sounds dumb. The Beatles taught us that with basic, boom-chick songs like 'When I'm Sixty-four.' A song played with feeling and integrity becomes appealing."

Regardless of the aesthetic level of the music, the basic idea should be clearly defined, either as specific notes spelling out precisely what to play, or as an attitude written in words. Jingle sessions don't afford the luxury of spending hours and hours trying out different parts. Most of those decisions should be made before the date begins. "Our sessions don't have a party atmosphere at all, or come across as laid-back and loose," says McFaul. "It's grown-up men and women taking care of business in an artistic vein. Nobody here has the attitude that 'This is *only* a jingle.' "

Production Sessions

At the start of the multitrack recording date, the writer of the tune plays the music on the piano, or plays a demo for the musicians, and supplies them with a couple of basic remarks about what the jingle is supposed to be. Then he counts out the groove, and lets the players interpret their parts just to see what happens. "Sometimes the guys in the rhythm section do something totally unexpected," McFaul recalls, "and we'll find something that works better than we imagined. Then we can shape it, and refine it."

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Because the song is on paper in a sketchy form, it's adaptable, adjustable, and shapeable within that first hour. Once the rhythm section leaves, however, the horn players and string players called in to overdub subsequent parts stick closely to the original charts.

"You're dealing with an intangible feeling in the song's foundation," says Lucas. "The structure, melody, or chords don't change — only the input or attitude given by each player's imagination. There's a certain pliability to the rhythm section that doesn't exist with the horns and strings."

There is a limit, of course. Surprises or drastic left turns on bass, for instance, don't work, because there are too many people involved in the overall recording process. In addition, the agency representative expects to get something resembling the demo he commissioned.

"You're dealing with sophisticated people who know what you're talking about when you describe how you'll add the strings and horns," says Lucas. "The changes during a date shouldn't be enormous, but rather small, pleasant surprises. You really have to know the end result before going into the studio."

Like musicians, jingle singers are called in on a per-session basis, and chosen for their particular versatility and skills. According to Lucas and McFaul, they rely on a core group of about 30 New York City-based vocalists that are all ace vocal arrangers, and

The Warehouse control room houses a Trident Series 80 console, Ampex MM-1200 multitrack and ATR-100 mastering machines, plus a full rack of outboard equipment.

ultra-pro readers that hear and sing well.

To demonstrate their prowess, Lucas describes the demands of a typical vocal-overdub session: "We hire eight people to sing the melody three times; that's 24 singers. Then they harmonize three times; that's 48 voices. A couple of extra vocal parts here and there brings the total to approximately 64 vocal performances on a single jingle. Yet when you listen back, it sounds like eight people singing once. That takes an amazing amount of speciality in terms of concentration, knowing when to cut-off, diction, turn of words, and blend. Jingle singers are probably as close to the best musicians you'll encounter during the day, and that includes the rhythm section, horns, and strings."

Engineering Pointers

For the engineer as well as the musicians and producer, speed is the most important consideration when committing a jingle to tape. Unlike a record date, the control room contains not only musicians and other studio personnel, but usually representatives from the client company and advertising agency executives. Every effort therefore, must be exerted to accomplish the most in the shortest amount of time. The keyword here is "preparation."

From the beginning, independent engineer Michael DeLugg strove to develop a close working relationship with Lucas and McFaul, which means entering a project while it is still in the "pre-pre-planning" stages. "Tom will play me a song and ask, 'What do you think of this for such-and-such a product,' "Lucas explains. "We talk about

the music, and I tell him what I like and don't like. By the time the client walks in the door, I know what's been written, what the client prefers, the time frames for all the recording sessions, the length of the spot, what the instrumentation is, and a rough idea about how many tracks and effects I'll need. I also try to discuss the chart with the arranger before we get under way. And, because I read music, I can follow the chart as we're going.

"Knowing so much about what's going on makes me feel more comfortable, and I usually don't have to ask too many questions during the session. We end up working faster; people walk out happier; and when the jingle goes on the air, it sounds better."

To make matters even easier, The Warehouse 24-track studio owns a full complement of house instruments: drums (a house kit, and a Linn Drum Computer); keyboards (grand piano, Fender Rhodes, Sequential Circuits Prophet, Oberheim OBX, Yamaha GS-2, etc.); instrument amps and several extra guitars.

"If a player comes in with just his electric guitar and we decide that an acoustic would be better, we're covered," says DeLugg. "Plus, I'm so familiar with the sounds of our instruments that I know all the ways to mike them for their particular characteristics."

DeLugg finds that he mikes instruments in pretty much the same way regardless of whether he's doing a jingle session or a traditional record date. "I try not to make a big deal out of what I'm doing technically. I find it's better to let more of the energy and 'magic' come from the players. I may use the extra time on a record date to experiment a little more. Or, if I'm dealing with a ste-

reo product, I need to be a little more concerned about the placement."

Many commercials get mixed down to stereo for FM radio broadcast, or for agency presentations to clients. If, for example, an agency is putting on a demonstration for Pepsi Cola in a large, living-room environment, or small theater with large speakers, DeLugg takes five more minutes to develop the stereo perspective; the result, he considers, sounds sufficiently more impressive.

Track assignments generally remain about the same, too. In the case of TV commercials, SMPTE timecode locks the audio and video work tapes together, and gets placed on track #24. A click track, two tracks for Valley People Fadex console automation data, and a 60-Hz video sync track eat up four more channels, which leaves only 19 open tracks for music recording.

"There's been no problem with the 60-Hz pulse [track #23] sitting between the high-frequency signals like SMPTE [track #24] and the data [tracks #22 and #20]. A music channel usually gets recorded between the two data tracks to avoid data crosstalk. The click goes on track #1, an edge track, so it's convenient to turn on and off in the monitors and headphones.

The band members get a cue send that resembles the control-room mix, the major difference being a click track that's sent to the players, but taken out

of the playback monitors. If ech[o/]reverb is being added to the final mix the control room, the musicians wil receive a reduced amount in their cue system, to keep their headphone signals clear and easy to understand. For musicians such as the trumpet players, however, DeLugg uses additional echo to create the illusion of a bigger room, so they don't have to overblow to get their sound across. "If the headphones make the trumpets sound like they're tight-miked," he explains, "the guys have trouble playing, and we don't get them recorded in the hour we have scheduled."

By the time the band runs down the spot once or twice, DeLugg has his sounds set, and is ready to start recording. He tries to record some kind of rough mix on tape as quickly as he can, so that the rhythm section, or whoever is out in the studio, can come into the control room, listen to the track, and discuss the overall blend. That first recording also provided a reference they can check against the video tape to ensure that all the timings are correct.

DeLugg will record most of the vocal work in ensembles of five to eight singers, with one mike for the girls and another for the guys. But 60-second commercials require a lot of solo work when there's a story to be told musically.

"We start with piano, vocal and simple orchestration, and build it all up like a mini record," he explains. "That

ss track problems. I try to touch with the arrangers s, so I always have plenty en. Sometimes it means live bouncing if we double-k the vocals [recording the i of a previous track with a live vocal], because that saves bouncing time and tracks right away, and we can get on with the harmonies."

DeLugg doesn't like to record effects on the basic tracks, although occasionally he gets a track with a delay, phasing, or pitch-shifting device that is so appropriate, or so much a part of the sound, that he'll lay them down live. "But I calmly inform everyone that this is what they are now locked into," he

laughs. "If I have to do a remix for some reason in three days, I'd rather have as little outboard gear as possible to set up, so I can get that remix done quickly.

Generally speaking, DeLugg says that he prefers to add the required effects during the mix, when all the musicians and singers have gone home. "When we take a little break to clear our ears before we mix, I'll turn to my assistant and let him know what gear I'll need. After the break, he has all that patched up and ready to go."

Mixing for the Medium

Where the commercial is destined to play — whether on a TV speaker, 35mm optical, 35mm mag, videotape, or radio

— determines how DeLugg mixes the soundtrack. Jingles for radio campaigns only might need to be in stereo for FM stations. In that case, he goes for a broader, fuller bandwidth spectrum, and doesn't "pinch" the tracks quite as much.

The restricted bandwidth of an optical soundtrack is another story, however. For truly accurate representation, DeLugg feels that it's important to mix on monitor speakers that reflect what the eventual outcome will be. He starts with a set of medium-sized bookshelf speakers that are just loud enough to give him the "kick" he needs to get a good mix. Then, to fine tune the mix, he slowly steps his way down through small pairs until he arrives at the one installed in the studio's television set.

"It really depends on the commercial, the arrangement, and the song as to what is important, and what needs to come through. If I see there's a string line being lost, I may switch to Auratone [Sound Cubes], let's say, to find out why, and then make corrections using either EQ, a change in level, or whatever. It's never simple, but you have to figure out a solution quickly."

Lucas and McFaul sessions usually host a crowd of five or six "big-wigs" sitting in on the mix. According to DeLugg, "they all think mixing is easy — 'Just push up the faders and you got it.' When the arranger says, 'More strings right here,' I have to assess what he means by that, and translate that to some mechanical action. If I think there's enough strings, and more will get us into trouble — like masking the vocal, or a nice piano lick they've wanted louder for the last hour — I have to let them know why. You have to be a little bit of a psychologist when you have five or six producers sitting in on a mix!"

The biggest mixing consideration, DeLugg offers, is maintaining control over the various instrument sections, such as trumpets, trombones, and french horns. "If a client feels the french horns state their product very well, I better have them on their own track,

Michael DeLugg started his engineering career at A&R Recording Studios in New York City as an assistant engineer for people like Phil Ramone and Donny Hahn. He moved to Media Sound Studios, where initially he worked primarily on jingles. But, by hanging out at the studio all the time, he picked up extra dates as a first engineer, and became so familiar with the clients that they began to ask for him by name. Record work with artists such as Brownsville Station, the Isley Brothers, Van McCoy, The Stylistics, and Barry Manilow soon followed. As the record business started to slow down, DeLugg decided the time was right to freelance after nine years at Media Sound. He returned to jingle work and, two years later, settled down at Lucas/ McFaul's in-house 24-track studio, The Warehouse, where he works steadily as an independent engineer, with an occasional record date for Barry Manilowe.

which may mean putting the trumpets and trombones together. Or similarly, I may have a client who is 'violin happy.' In a situation like that, I'll put the violins on one track, and the violas and celli together on another track, so there's enough flexibility for me to make the client happy. Only if we're eventually going for a lush stereo [mix] can I record the strings in a nice broad live stereo, and not fool with violins here, cellos there."

With speed being of paramount importance, a few state-of-the-art tools, including console automation to memorize fader-level and mute information, make mixing more foolproof. DeLugg spends a couple of minutes adjusting the music mix without looking at the video accompaniment. Via SMPTE timecode interlock he can play back the audio mix data and video in perfect sync. Then DeLugg can watch the picture without worrying about duplicating a complicated mix in real time.

"When I first came to work at The Warehouse, I was saying 'Live mixing is better.' But with four or five producers from an agency all giving commands, being able to fine tune tracks with just a touch of a fader in an isolated update mode saves a *lot* of headaches. Plus, I know the whole mix will play back the way I set it the first time by reading back the automation data, so I can look at the television monitor, too, and see what's going on. When somebody says,

'I'm having a problem where the girl kicks the football,' I know what he's talking about, because I've seen it."

SMPTE timecode also comes in handy when too many music parts necessitates a doubling up of tracks. For instance: a last-minute sax solo that occurs only at the beginning may share a vocal track that's only used during the last 12 seconds of the jingle. Using console automation, DeLugg can assign that track to two separate faders. Labeling one the "sax" and the other the "vocal" fader, he sets the appropriate EQ, echo, and other effects on each channel, and teaches the computers the level moves.

DeLugg tries to mix jingles very simply, so that there aren't too many surprises. "If a jingle suddenly starts distorting, or pumping the limiters at a radio or TV station, because the changes in level are too drastic, the commercial will sound bad on the air," he offers. "If I need a little bit of compression on the overall mix, or a little riding on the overall level, I'll do that before the product goes out the door."

A Specialist Art

With all the talk about hooks, knowing how to write for specific markets, having such technically proficient people and equipment at their disposal for writing, producing, engineering and recording, why don't Lucas and McFaul take on the lucrative world of Top 40 music?

"That's another craft," answers David Lucas. "I did that for a time. I have a couple of Gold records on the wall, but I didn't enjoy it. I think there is more 'giving up' on projects in the record business, because you have to consider the trends, consumers, that you're writing for 14-year-old girls, maybe college kids, when you write records. Records are much more specific than jingles.

"We actually have more control in advertising than songwriters do in the record business, even though we're selling products. This way we hire the top studio musicians in the world to play what we want them to play, and they're gone in an hour. We start a job at 10 a.m., and it's finished that night. When you do records, it's a complete life-style commitment."

Tom McFaul's opinions follow a similar pattern: "If you get on a bad record project, you're stuck there for four weeks, or whatever. If it is a bad jingle, it'll be over in an hour. Jingles allow us to dabble in all the different pop styles. Away from the commercial business, I get to work on something like a piano sonata. I'm not thinking at all about publishing it, because I do for myself.

"Jingles are more fun than anything else we ever thought we'd be doing," he continues enthusiastically. "We're also making a good living. We're not losing our writing chops, and the people are nice. What more could we want?" ■■■

YAMAHA LAUNCHES REV1 DIGITAL REVERBERATOR

According to Yamaha engineers, the REV1 offers more extensive, precise and easily understood control of sound than any previously available digital reverb/delay system or mechanical reverb devices, all of which are limited in the number of controllable sound parameters.

With the REV1, the user has control over such variables as: sound directly from the source; "early reflections" bouncing off surfaces; as well as "subsequent reverberation" in which the reflections are multiplied. Total electronic control can be maintained to compensate for, or simulate the acoustic properties of any environment, accounting for such variables as room size, room shape, contents, acoustic absorption coefficient, and listening positions.

The REV1 can create and control up to 40 early reflections, and provide up to 99 seconds of subsequent reverberation. Even the relative timing between early reflections, their absolute level, and when they begin, is controllable. Users can adjust the delay and reverberation independently and, as with the early reflections, the delay before onset of reverberation can be precisely set, as can the reverberation density and absolute level. The liveness of sound can be altered by changing the level of "later" early reflections, relative to the "early" early reflections.

Reverberation time can also be adjusted differently in each of the four frequency bands, and the crossover frequencies are selectable. Once the ratio of RT60 in the various frequency bands have been set, "basic" reverberation time can be changed with a single adjustment. The other three bands then automatically track the midband RT60, if desired, to save adjustment time and retain a given sound quality.

The front panel of the 19-inch rack-mountable main unit included basic controls for recalling any of 30 preset and 60 user-programmable effects, and modifying them to some degree. The remote control unit includes 64 pushbuttons (most of which are illuminated), 10 control knobs, LED numeric display windows, twin 16-LED level displays, and a high-resolution LCD window that graphically depicts adjustments as they are being made.

Once a desired effect is achieved, it can be stored in any of the 60 memories for instant recall, and there are 30 additional memories for factory preset sounds. All 90

can be completely edited. LEDs on the remote control panel reflect the set value or indicate how to move the knob to match the value of the recalled sound.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134
For additional information circle #97

AUDITRONICS 300 SERIES PRODUCTION CONSOLE

Representing a new concept in operational control, the 300 Series is designed for four- or eight-track production with sub-mastering to stereo and/or mono, and features up to 32 mono or stereo inputs with or without equalization, auxilliary sends and returns.

Monitoring facilities including cue and stereo solo; a complete user-programmable internal and external Logic System; remote control and interface capability to external switchers or editors for Audio-Follow-Video; and a wide variety of options and accessories for signal processing, routing, or user customization.

AUDITRONICS, INC.
3750 OLD GETWELL ROAD
MEMPHIS, TN 38118
(901) 362-1350
For additional information circle #98

ALL-HORN-LOADED CONCERT SOUND SYSTEM NEW FROM EV

The basic EV portable, horn-loaded concert sound system consists of a single TL4025 subwoofer, TL1225 midbass horn, and appropriate EV HF constant-directivity horn. It is possible — and usually desirable — to create more massive systems by using multiples of each of the basic components, stacking them vertically so as to maintain a horizontal coverage angle of 60 degrees at critical mid and high frequencies.

The TL4025 folded horn bass system, outfitted with a 400-watt EVM-15L Pro-Line driver, provides maximum low-frequency output down to 40 Hz. Its unique "dual-S" folding scheme permits the use of a hyperbolic-exponential taper rate to elicit maximum LF output from the 168-pound horn. The TL4025's specifications

include a conversion efficiency of 21% and a 1-watt/1-meter sensitivity of 104 dB.
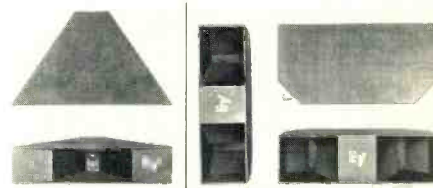
The TL1225 midbass horn with a single 300-watt EVM-12S Pro-Line driver operates in the range of 125 Hz to 1.25 kHz and, in addition, provides a constant-directivity 60-degree horizontal coverage angle that complements the HF horn operating above 1.25 kHz. A single TL1225 produces a quoted sound pressure level of 109 dB (1 watt at 1 meter), and has a conversion efficiency approaching 20%.

Enclosures are fabricated from 15-ply Baltic birch finished with a textured epoxy coating for improved ruggedness. Both enclosures come complete with hardware and measure 60 inches wide to facilitate proper array stacking.

The new TL4025 and TL1225 horns have suggested list prices of $998 and $798, respectively.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831
For additional information circle #99

NEW REMOTE CONTROL SYSTEM FOR PCM-3324 DIGITAL MULTITRACK FROM SONY

The RM-3310 remote provides remote capability of the PCM-3324's recording functions, as well as synchronization of additional multitracks. Simple interface of two transports creates a 48-channel digital recording system. The new unit is designed for synchronizing up to 15 multitracks, and will be compatible with future Sony digital multitracks.

The RM-3310 is comprised of two modules — a rack-mounted audio control unit and a system control unit that either can be rack-mounted or operated on a desk top. The system command remotely controls all transport modes, plus Rehearse and Editstop, and shows function mode switches on all channels and status indi-

cation. The autolocate functions include location of cue points, location with variable pre-roll, and automatic repeat, speeding punch-ins and electronic edits.

After auto punch-in and punch-out points are determined, the settings can be rehearsed and then accurately executed. Takes from separate tracks are then merged into one track through electronic crossfading of designated tracks. Edit functions normally performed on a single machine can now be performed on synchronized multiple machines. A 10-key pad and memory provide settings for up to 100 cue points.

**SONY PROFESSIONAL
AUDIO PRODUCTS
SONY DRIVE
PARK RIDGE, NY 07656
(201) 930-6432**

For additional information circle #101

ACO PACIFIC INTRODUCES ACM48 CONDENSER MIKES

The ACM48 family of studio microphones is available with five interchangeable capsules. Each mike is individually calibrated and supplied with a "Birth Certificate" documenting the 20 kHz response.

The ACP48 companion pre-amplifier is phantom-powered, and supplies a 200-ohm output impedance. Power may be supplied by ACO's portable supply, "The Ghost," or a conventional 48V phantom source.

A quoted 116 dB of dynamic range results from the low (18 dBA) noise floor, and the 134 dBSPL (0.5% THD) specifications.

The Ghost, a two-channel 48VDC portable power supply originally developed for use with the ACM48 mikes, permits shotgun and other high-quality microphones requiring phantom power to be used with VCRs, audio tape recorders, and low-cost consoles not providing 48-volt power.

A standard 9-volt alkaline battery provides a stable, clean source of regulated 48-volt power for up to 16 hours or more depending upon microphone requirements. A jack for external power from a wall adaptor, or even an automobile battery, is provided.

Manufacturer's suggested prices: ACM48 capsule $120; ACP48 pre-amplifier $280; and PS9048 "The Ghost" PSU $249.

**ACO PACIFIC, INC.
P.O. BOX 5506
SAN MATEO, CA 94402
(415) 591-8509**

For additional information circle #102

INTERFACE MODEL 324 MODULAR CONSOLE

The Series 324 incorporates all of the functions of the mixer in each module, so that, for example, a 28-input/24-track console requires simply the insertion of 28 input modules in the standard frame. Each module incorporates a track output master with an LED VU level indicator, as well as input module features including track assign, four equalizers, and four cue/effects sends.

Also included is the Model NA Monitor module, which makes three stereo mixdowns from the monitor send section of the individual modules; these can be used for control room, studio, and 'phones in a recording system, or house mix and operator's mix in a sound system. The NA module also includes talkback and echo

returns.

For recording, the system can be set up for up to 24 tracks, and provides a simultaneous stereo mixdown — a stereo mix for control room with solo and another mix for studio without solo. For sound systems, the console provides up to 24 submixes, each panned into the stereo house output, with another stereo output with solo functions for the operator. Many points in the mixer can be soloed into the monitor.

The Series 324 construction without meter bridge and with masters incorporated in modules results in the maxilum of features in the minimum of space and weight.

**INTERFACE ELECTRONICS
6710 ALDER
HOUSTON, TX 77081
(713) 660-0100**

For additional information circle #103

YAMAHA YDM2600 DIGITAL DELAY SYSTEM

The YDM2600 consists of two units: a main rack-mountable chassis with local control of many parameters, and a small hand-held remote control unit with even greater control capability. The remote features an alphanumeric LCD readout, while the main unit has a back-lit LCD. These displays indicate the selected delays, whether a given output is bypassed, and more.

Four balanced XLR inputs are provided, each with its own level control and four, 16-LED level meters. A stereo headphone monitor output facilitates setup and test-

ing, and the delayed audio outputs are accessed by eight balanced XLR connectors.

The unit offers three modes of operation: One-in Mode with eight discrete outputs at up to 2.66 seconds of delay for each; Two-in Mode, each with four discrete outputs at up to 1.33 seconds delay; and Four-in Mode, each with two discrete outputs at up to 0.655 seconds delay.

Entire setups may be stored in any of 12 user-programmable memories, four memories for each of the three operating modes. Additionally, provisions have been made for an additional 12 read-only memories for custom installations where system requirements are regularly changed between several "fixed" configurations.

Since the YDM2600 basic delayed output can be set to within 20.8 microseconds resolution, it should be particularly useful for synchronizing audio with video on a satellite up/down link, for example. In the case of disk mastering, the YDM2600 eliminates the need for preview heads.

The YDM2600 also can be controlled by computer utilizing an RS-232 interface and separate serial input/output connections, allowing automatic effects switching.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134

For additional information circle #105

MICRON TX-203 WIRELESS MICROPHONE AND RECEIVERS

The TX-203 hand-held mike employs an interchangeable head assembly, allowing its use as either an omnidirectional or cardioid pattern condenser microphone.

Also new are the MDR-3 single-channel, and MDS-2 four-channel space diversity receivers. The latter can operate up to eight diversity channels with antenna distribution, audio output, and universal power modules, all in one unit. Each channel is easily added to the frame, simply by plugging in.

MICRON AUDIO PRODUCTS
210 WESTLAKE DRIVE
VALHALLA, NY 10595
(914) 761-6520

For additional information circle #106

EXR UNVEILS PROJECTOR MODEL SPII PROCESSOR

The Projector™ is a two-channel psychoacoustic processor incorporating the same four psycho-acoustic process modes and mix/solo capabilities previously only available in the EX Series. The SPII utilizes pre-selective 180° phase notching, psycho-acoustic frequency juxtapositioning, time delays and pre-emphasis to precondition the audio signal to match the ear's hearing mechanisms.

The new unit was designed to be similar

in operation to an image enhancer used in video production. A video engineer would not boost blue (a visual frequency) to give clarity to a picture, because it would throw off the natural color relationships. According to EXR, using equalization to give clar-

ity and separation to audio material also throws off the natural tonal relationship of the signals. Instead, the Projector processes the whole audio frequency spectrum simultaneously, notching out distortion bands and centerpoint frequencies that coincide with the ears' limiters.

EXR's exclusive process then uses

octave-related delays and pre-emphasis to apparently replace the notched out narrow frequency bands, thereby maintaining the tonal integrity of the audio signal while achieving "dramatic clarity, separation, and intelligibility."

Suggested list price of the new EXR Model SPII Projector is $549.

EXR CORPORATION
3373 OAK KNOLL DRIVE
BRIGHTON, MI 48116
(3132) 227-6122

For additional information circle #107

NEW EV TWO-WAY SPEAKERS

The trio of two-way speaker systems intended for permanent sound reinforcement applications where a wide, controlled coverage angle and high efficiency are desired comprise the FR15-2, FR12-2, and PI100.

Citing numerous applications where just one system capable of uniform coverage over a wide frequency range is called for, Jim Long, director of marketing/professional sound reinforcement products adds, "Such performance is available from separate components —a wide-angle horn, compression driver, low-frequency system, and crossover network — but with three big drawbacks: relatively high cost; complex installation; and a clumsy 'utility' appearance."

Crossover frequency and speaker component geometries of the new loudspeakers have been carefully selected so that directional characteristics of the woofer and constant-directivity horn match at the crossover frequency to create a special system type: the constant-directivity system. And, unlike a conventional horn, an EV constant-directivity horn is said to maintain its rated beamwidth to the highest frequencies, thus assuring broad, uniform coverage without hot spots and dead spots.

Two models in the series, the FR15-2 and FR12-2, have oak-grain vinyl enclosures with detachable beige grillecloths. Suited to outdoor as well as indoor use, the weather-resistant enclosure of the PI100 is a molded polyethylene with a water-resistant grille. T-nuts are embedded in the sides of all cabinets to facilitate suspension.

Low frequencies of the FR15-2 are handled by a 15-inch EVM-15L Series II woofer mounted in a 4.3-cubic-foot optimally vented enclosure. Frequencies above the 1.5 kHz crossover point are handled by a DH1202 compression driver on a 90- by 40-degree constant-directivity horn.

The FR12-2 low-frequency section features a 12-inch woofer mounted in a vented 1.8-cubic-foot enclosure. Frequencies above the 1.5 kHz crossover point are handled by a 1½-inch Super-Dome tweeter coupled to a 9-inch Direktor. Except for its enclosure, the PI100 is identical to the FR12-2.

The new Electro-Voice speakers have the following pro user net prices: FR15-2 $665; FR12-2 $417; and PI100 $417.

**ELECTRO-VOICE, INC,
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831**

For additional information circle #109

LOW-COST R1000 DIGITAL REVERB FROM YAMAHA

Controls on the R1000 enable reverb and direct sound to be varied continuously in any of four different reverb time settings. A three-band parametric equalizer enables continuously variable adjustment at low, mid and high frequencies.

According to Yamaha, "the R1000 is an ideal, economical replacement for mechanical reverberation units and its light weight [10½ pounds] and small size [19 inch rack-mountable] eliminates the hassle of carting larger, more temperamental reverb units. In fact, because it is digital, the R1000 completely eliminates undesirable physical vibration effects of spring-type reverb units."

The R1000 will carry a suggested retail price of $795.

**YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134**

For additional information circle #110

AUDIO ENGINEERING ASSOCIATES NOW OFFERING COLES 4038 RIBBON MIKE

The current Coles type 4038 is the culmination of 30 years of development and usage by organizations such as the BBC. The best ribbon microphones have been renowned for their warmth, good response, and smooth, mellow, extended top-end; liabilities have been size, weight, and fragility. The 4038 is described as being the modern solution to these problems.

Ribbon material is a carefully selected pure aluminum foil, corrugated and precisely tensioned between high permeability pole pieces. Slightly magnetic woven magnetic screen mounted each side of the ribbon provide a precise degree of damping, and act as stops to prevent overstressing of the ribbon under accidental wind blasts.

The ribbon combines the functions of a very low mass, critically-damped acoustical diaphragm, and a low resistance half-turn dynamic coil. The ribbon and pole piece shapes have been carefully optimised for a flat frequency response with a higher output efficiency than earlier models. The moving mass of the ribbon is only about 1/500th that of most dynamic moving coil microphone systems and, having great flexibility, the ribbon is easily tuned to a very low basic frequency. Combined with correct acoustical damping, these features are said to give the microphone a very flat and extended bass response.

All microphones tend to suffer a change in polar response at high frequencies, due to their physical size compared to the sound wavelength. The 4038 ribbon and

pole piece system is completely symmetrical in the horizontal plane. This, combined with the short vertical length of the ribbon, and the acoustically compensating design of the casing, help to maintain a constant shape of the polar response of both planes.

AUDIO ENGINEERING ASSOCIATES
1029 N. ALLEN AVENUE
PASADENA, CA 91104
(213) 798-9127

SONY INTRODUCES MODULAR SYSTEM FOR MULTIPLE COMPACT DISC PLAYERS

The new CD Modular System consists of the CDS-3000 control unit and CDP-3000 Compact Disc player(s). The new CDP-3000 is a variation on the CDP-5000, while the CDS-3000 control unit has been designed for programming of Compact Discs in audio production applications, as well as in radio and TV stations.

The CDS-3000 is capable of controlling two CD drives either by program or by manual operation. Accurate cueing is made possible through the aid of a 10-key pad, and a rotational search dial. Up to eight programs can be handled simultaneously and then played back consecutively, with one-frame accuracy at start/stop points.

The units can be mounted in a standard 19-inch rack, and all operation is accomplished automatically. The programming of two CD players allows for smooth and accurate segueing from different source materials for uninterrupted production or airplay requirements.

SONY COMMUNICATIONS PRODUCTS COMPANY
SONY DRIVE
PARK RIDGE, NJ 07656
(201) 930-6432

JBL INTRODUCES MODEL 2386 FLAT FRONT BI-RADIAL HORN

Providing uniform on- and off-axis frequency response (from 500 Hz to 16 kHz in the horizontal plane, and 2 kHz to 16 kHz in the vertical plane), the 2386 horn has a nominal 40- by 20-degree coverage angle. The unit has been designed for flush cabinet mounting, or compact cluster installations.

The highly dependable performance of the new flat-front 2386 is said to greatly simplify cluster design, minimize the need for horn overlapping, and virtually eliminate lobing and comb filter effects. Computer-aided techniques were employed to derive the horn contours in the horizontal and vertical planes, to yield smooth

response, low distortion and even coverage, avoiding the performance disadvantages of sharp flare transitions and flat sidewalls.

The new model is constructed of injection-molded, reinforced polyurethane, for light weight, superior strength and

freedom from resonances. The horn's small vertical mouth dimension (slightly larger than the compression driver used to drive the horn) allows compact single and multiple horn/driver systems to be assembled. Should vertical pattern control be needed below 2 kHz, two or more horns may be stacked vertically to restore full Bi-Radial performance.

The 2386 will accept JBL's two-inch diameter 2441, 2445, or 2482 compression drivers. With the addition of the 2327 adaptor, the horn also will accept the one-inch throat 2425 driver.

JBL, INC.
8500 BALBOA BLVD
NORTHRIDGE, CA 91329
(213) 893-8411

FOSTEX B-16 16-TRACK ON HALF-INCH MACHINE

Described as the first commercially available 16-track recorder/reproducer using half-inch tape, the B-16 offers built-in Dolby C noise reduction as standard. Because of its compact size (approximately 17 by 17 by 9 inches deep, and weighing 66 pounds, the B-16 is said to be ideal for cramped remote trucks, control rooms, or spare rooms.

The transport features a 3-motor design with two direct-drive reel motors, and a servo-controlled capstan motor. Running speed is 15 IPS with ±15% variable operation in both record and reproduce modes. A real-time tape counter operates with search-to-cue function from any mode.

The B-16 is video interlock ready — a multi-pin connector (rear panel) is pre-wired for SMPTE interlock synchronizers such as BTX, Adam-Smith, Q-Lock, CMX and EECO. The machine is delivered with pin-out information and mating connector for fast, accurate interface.

The electronics package includes individual record/reproduce cards for each channel (adjustments are easily accessible behind the removable meter panel); LED bar graph metering system with peak ballistics on attack and VU ballistics on decay; full frequency response in sync mode (standard model has two heads); and multiple track punch-in/out with one button operation.

The basic B-16 with belt-drive capstan has a recommended retail price of $5,900; optional direct-drive PLL capstan costs an additional $700.

**FOSTEX CORP OF AMERICA
15431 BLACKBURN AVENUE
NORWALK, CA 90650
(213) 921-1112**

For additional information circle #117

MOSFET 900 POWER AMP FROM BANNER

The new amplifier utilizes power MOSFETs, and features power output ratings of 300 watts RMS per channel into 8 ohms, and 450 watts into 4 ohm loads. In the mono bridging mode the rated power is 900 watts into 8 ohms. A dual-speed, thermostatically controlled fan keeps the 900 running cool under heavy concert use, and quiet enough for a studio control room.

It has front-panel level controls, LED meters and an exclusive new stereo limiter. The meters can be front panel calibrated to a continuous range of output from 56 to 450 watts RMS into 4 ohms; this calibration point is the zero reference point of the meter, and is automatically the limiter threshold point. The limiter slope may be varied from soft to hard limiting action, and is fully defeatable.

All controls are front panel operated, including circuit breakers and input impedence selection. Active balanced inputs on XLR and ¼-inch connectors are standard. The outputs have full speaker protection, including DC fault sensing. Distortion typically ranges from 0.005% to 0.2% depending on power and frequency.

The suggested list price of the MOSFET 900 is $1,250.

**BANNER
P.O. DRAWER 1803
SHELBY, NC 28150
(704) 487-7012**

For additional information circle #119

MODEL AM-1 PHASE METER FROM B & B SYSTEMS

A new stereo audio and timecode phase verification system has been introduced by B & B Systems, a recently formed company set up by William Burnsed and John Bradford.

The AM-1 combines an X-Y 'scope, average-reading VU meters, peak-reading LED meters, plus timecode oscilloscope display, in a single chassis.

The X-Y 'scope displays stereo phase and separation, while the field-locked 'scope shows timecode phase and genlock. The unit also features separate peak and average audio level meters, and 'scope display of left and right audio channels.

According to Burnsed, "The product is ideal for post-production, television field production, broadcast stations, or any place where stereo audio or timecode is used."

The AM-1 sells for $2,800.

**B & B SYSTEMS
28111 N. AVE. STANFORD
VALENCIA, CA 91355
(805) 257-4853**

For additional information circle #120

SOUNDTRACS 16-8-16 RECORDING CONSOLE

Innovative features of the 16-8-16 include eight additional inputs in remix or record, with equalization and fader reverse to control them. In addition, it is now possible to have 16-track monitoring with simultaneous recording. In all the mixer

offers three permutations of input/output: 32-2, 24-8-2, and 16-8-16.

The 16 tape returns are normalized to the 16 monitor sections, so no replugging is necessary for remix. Visual level monitoring is available on an 18-way meter bridge, each containing a 10 element, bi-color LED ladder that actually correlate directly to the line of controls being used.
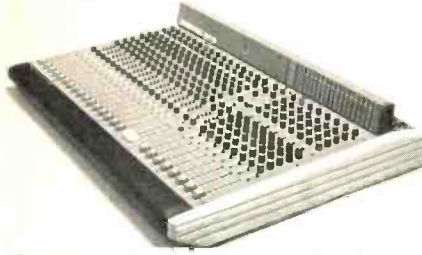
Power to the unit is provided by an external free-standing unit with switchable mains input (100 to 240 volts) and master phantom on/off switch.

Recommended retail price of the 16-8-16 console is $6,595.

SOUNDTRACS
262A EASTERN PARKWAY
FARMINGDALE, NY 11735
(516) 249-3669

PRIMUS MIKE/LINE MIXERS FROM RAMKO

The new P-4M and P-5MX mixers are available in mono and stereo versions, in table-top or rack-mount configurations.

The P-4M provides four channels and six balanced inputs, with selectable Hi/Lo shelving EQ for channels #1, #2 or all. Other features are selectable peak/VU meter ballistics, headphone driver, 'phone, master and monitor controls, and cue on all inputs.
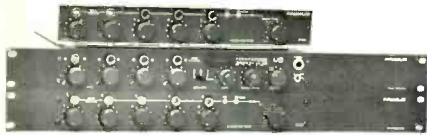
The P-5M is designed to function as both an expander for the P-4M (to provide a total of 11 inputs and nine channels), and as a stand alone five-channel mixer with send/receive on each channel.

Both units feature XLR-type connectors, balanced inputs and outputs, gain select on all inputs (mike thru +26 dBm). Conductive plastic controls and long-life switches also are featured.

RAMKO RESEARCH
11355-A FOLSOM BLVD
RANCHO CORDOVA, CA 95670
(916) 635-3600

FREEDOMIKE WIRELESS SYSTEM BY LECTROSONICS

The Performer wireless system features a transmitter that plugs on to any professional microphone or to any high-impedance instrument pickup. Transmitter audio gain is easily adjusted to match the individual voice, microphone, and instrument. The transmitter also features an internal antenna to eliminate the dangling wire or protruding whip common to most wireless microphones. The receiver is

equipped with balanced line outputs for connection to any sound equipment.

The new system uses both compression and compander/expander techniques to ensure distortion-free performance over the broadest range of audio input levels, and is available on five protected frequencies to ensure interference free operation.

The Performer system retails for $689.

LECTROSONICS, INC
P.O. BOX 12617
ALBUQUERQUE, NM 87195
(505) 831-1010

IBANEZ INTRODUCES RANGE OF GRAPHIC EQUALIZERS

The GE1502 dual 2/3-octave and GE 3101 third-octave graphics occupy a single

rack space, and feature EQ in/out, switchable high-pass three-pole rumble filter for PA applications. The range of boost and cut is selectable between ±6 dB for subtle EQ curves, and ±12 dB for more extreme control. LED's indicate all switched functions and channel overload.

Specifications include frequency response: 20 Hz to 20 kHz, ±0.5 dB; TMD: less than 0.02%; hum and noise: less than −95 dBm (IHF-A); and maximum signal level (input, output): +20 dBm.
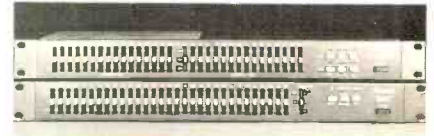
Suggested retail price of the GE1502 and GE3101 is $325.

HOSHINO (USA) INC
1716 WINCHESTER ROAD
BENSALEM, PA 19020
(215) 638-8670

OBERHEIM UPDATES DMX DRUM MACHINE

The DMX has been expanded with new software, and more than double the memory capacity. The new software allows for over 45 new features including 5,000+ event internal programming capacity; 200 sequence patterns; 100 songs; programmable tempo displayed in frames per beat; song and sequence length displayable in minutes and seconds; and selective cassette interface. . . . continued overleaf —

millisecond increments. With the SDE-3000's eight-channel memory capability, the user has footswitch assess to eight different effects. LED readouts accurately display all vital system functions.

The cost-effective SDE-1000 offers delay times of up to 1.125 seconds, and four channels of programmable preset memory. It features the same precision facilities for setting delay times, and also incorporates LED readouts.

Both the SDE-3000 and -1000 are equipped with four remote switch jacks for ease of operation: Delay On/Off, Hold (for repeating delay endlessly with adjustable tempo), Playmate (to set delay times remotely during performance), and Preset (for switching between memory channels). The SDE-1000 is also equipped with a modulation foot control jack for footpedal control of modulation rates.

ROLANDCORP US
7200 DOMINION CIRCLE
LOS ANGELES, CA 90040
(213) 685-5141

For additional information circle #130

BAG END INTROUDCES NEW STUDIO EFFECTS RACK

The new Studio Rack Model A contains 12 vertical rack spaces, two diagonally positioned spaces, and four spaces in the lower body that allow the outboard gear to point straight up at the user.

The rack is constructed of either oiled birch veneer, or a plastic laminated material available in a wide selection of colors. Custom-made versions can also be ordered to fit virtually any application.

Measuring 38 inches tall by 20½ inches wide, the Studio Rack is 29¼ inches deep at its deepest point, and comes complete with casters and a removable vented back that provides easy access to inner connections. At the bottom is an opening for wires, and a storage area for extra jacks, cable, and other items.

MODULAR SOUND SYSTEMS
P.O. BOX 488
BARRINGTON, IL 60010
(312) 382-4550

For additional information circle #131

EXR EXCITER MODEL EXIV

The new, fourth generation Model EXIV psycho-acoustic enhancer is an evolution in design from the EX Series. While maintaining all of the characteristics of the

EXIII, a totally new enhancement process, the new "A" mode, has been engineered to enhance and clarify the bass and lower-mid frequencies.

New features include: Sweepable frequency centerpoint control; process noise gate with threshold and release speed controls to eliminate background noise during quiet passages; adjustable process limiter to prevent high-frequency splash on dense program material; 0 dB or -20 dB switchable inputs/ outputs; and a peak-level switch which gives narrow- or wide-band enhancement.

Suggested retail price for the EXR Exciter Model EXIV is $1,690.

EXR CORPORATION
3373 OAK KNOLL DRIVE
BRIGHTON, MI 48116
(313) 227-6122

For additional information circle #132

NEW PORTABLE MIXER FROM SONY

The MX-P61 12-channel multipurpose portable mixer is small enough to fit into a standard 19-inch rack, and features all-transformerless balanced input and output circuitry for optimal signal clarity. Each of its 12 input channels is equipped with Cannon-type connectors for either microphone or line inputs. Four line and three auxilliary outputs are provided, together with comprehensive monitoring and talkback sections.

The unit also features three-band equalizers on each input, choice of VU metering or LED peak-program meters, high and lowpass filters, two stereo output limiters and choice of AC or 12-volt DC power operation.

The MX-P61 also offers selectable 48V or 12V AB phantom power, and selectable line/auxiliary output reference level of +4, +6, +8 dBm. The mixer weighs just over 40 pounds, and measures 17-inches wide, by 22⅛-inches deep, by 5⅛-inches high.

**SONY PROFESSIONAL
AUDIO PRODUCTS
SONY DRIVE
PARK RIDGE, NY 07656
(201) 930-6432**

For additional information circle #134

HAFLER INTRODUCES P225 POWER AMP

The P225 is conservatively rated at 175 watts per channel into 4 ohms, with less than 0.03% THD over the frequency range from 20 Hz to 20 kHz. The unit uses a puch-pull complementary symmetry circuit that employs MOSFET output devices and premium components throughout. Because MOSFETS are inherently self-protecting, the P225 has no need for complex and sonically degrading current limiting circuitry, the company claims.

The P225 employs rear-panel output fuses for load protection, as well as thermal circuit breakers mounted to the heat sinks. The thermal breakers will shut the amplifier down in the unlikely event it overheats. The unit can easily be converted to a monophonic amplifier by its internal mono/stereo switch; in the mono mode, the P225 will deliver 350 watts to an 8 ohm load. Input gain controls on the amplifier's rear panel make level matching possible with a simple screw driver adjustment.

Tentative prices of the new P225 amplifier are $499.95 fuly assembled; and $425.95 partially assembled.

**THE DAVID HAFLER COMPANY
5910 CRESCENT BLVD.
PENNSAUKEN, NJ 08109
(609) 662-6355**

For additional information circle #135

AMPEX MM-1200 MODIFICATION KIT FROM SYE MITCHELL SOUND

The computer-optimized modification kit, developed by Peter Butt, is intended to improve the audible performance of the MM-1200 multitrack. Modifications included in the user-installed kit are said to improve the reliability of the capstan servo system; the single-point Search-to-Cue system; enhance audio channel transient and frequency response; and reduce audio distortion, especially in the low-frequency region. The time required for record mode entry and exit also is reduced, along with reduction in obtrusive "thumps" resulting from bias ramping.

The kit includes all required parts of approved types and quality levels, along with specific installation instructions in a step-by-step format. Illustrations of part locations for the Audio Switching card, Reproduce Amplifier, and Search-to-Cue card are provided. Only normal electronics hand tools are required for installation.

Pricing for the kits is $1,200 for the 24-track kit, and $800 for the 16-track.

**SYE MITCHELL SOUND CO.
22301 CASS AVENUE
WOODLAND HILLS, CA 91364
(213) 348-4977**

For additional information circle #136

Q.SOFT-CONFORM SOFTWARE FROM AUDIO KINETICS

The new software package for the Q.LOCK 3.10C synchronizer is specifically designed to streamline the process of editing an original master audio track to an edited videotape. Simple key sequences automatically cue all the machines to the required points, drop into record when desired, and relocate for a review of the edit, also calculating offsets automatically.

Q.SOFT-CONFORM is normally configured for three-machine operation, with the facility for locking together the video machine and multitrack for ease of use. The original audio is transferred from the third machine onto the multitrack, and relay closures within Q.LOCK permit alternate track laying to be performed with an overlap between the segments for full mix/edit facilities.

When used in a two-machine configuration, Q.SOFT-CONFORM will enable two video machines to be used as a simple assembly video-editor, complete with color framing calculations modifying the machine offsets.

Previously, Q.LOCK interfaces for the Sony 5000 Series of U-Matic VCRs did not permit its use as a slave to an audio or

video master. The latest generation of interfaces for these machines is now capable of being used as master or slave. Existing Q.LOCK systems are retrofittable with the new interface.

AUDIO KINETICS INC
4721 LAUREL CANYON BLVD
NORTH HOLLYWOOD, CA 91607
(213) 980-5717

JBL MODEL 4645
SUBWOOFER SYSTEM

Developed to augment low-frequency reproduction in a variety of professional applications, including recording studios, motion picture theaters and sound reinforcement systems, the Model 4645's 18-inch LF component features a four-inch diameter voice coil and SFG (Symmetrical Field Geometry) magnetic structure for accurate, extended, deep bass response with minimal distortion. The driver is housed in a direct radiator, bass reflex enclosure constructed of dense stock, and braced for maximum durability.
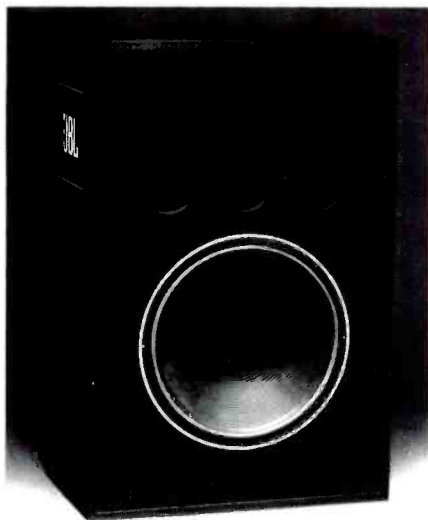
Tuned to 30 Hz, the enclosure has a net internal volume of 8 cubic feet. System response is said to be flat to 35 Hz, and usable to 25 Hz. For increased efficiency and extended response, multiple modules may be utilized. Use of JBL's 5234A electronic frequency dividing network with two JBL 51-5138 (80 Hz, 18 dB per octave slope) crossover cards is recommended to ensure optimum performance.

JBL, INC.
8500 BALBOA BLVD.
NORTHRIDGE, CA 91329
(213) 893-8411

SPECIAL MODIFICATIONS NOW OFFERED ON REVOX PR99

The eight special modifications for the Revox PR99 Reproduce Only tape deck are intended to allow "custom tailoring" to meet specific user requests in a variety of broadcast, industrial, and commercial background music applications.

The modifications available are: *low speed* (1⅞/3¾ IPS) in ½- or ¼-track formats; *Y-ready* signal to indicate tape is loaded but not moving; *Auto Cue* for automatic cueing to 25 Hz tone; *Auto Reverse* for continuous playback of ¼-track tapes in both directions; *Auto Repeat* for auto-

matic rewind and continuous playback; *capstan motor off* to disable capstan motor when clear leader or end of tape is detected; *mono* for playback of full-track tapes; and *quarter track* for playback of either ½- or ¼-track tapes.

In some cases, combinations of the above modifications are possible. For further information on multiple modifications and other modifications, please contact G.T. Sanford III:

STUDER REVOX AMERICA, INC
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

SHURE ADDS LAVALIER MODEL TO AUTOMATIC MIKE SYSTEM

The AMS28 condenser microphone is a lavalier unit that performs the same automatic functions as Shure's other AMS microphones, the low-profile AMS22 and probe-style AMS26. When used in conjunction with an AMS mixer, the AMS28 will turn on and off automatically in response to the wearer's speech. In addition, the exclusive design of AMS mixers and microphones prevents the AMS28 from being activated by any undesirable sounds that originate outside the microphone's 120-degree front acceptance angle.

Sound sources that originate beyond this "window of acceptance" will not make the microphone turn on, regardless of their loudness. When a number of Shure AMS microphones are in use at the same time, each mike will operate independently in analyzing its own soundfield, and determining whether or not a sound source is

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New Products

within the front acceptance angle.

When an AMS28 is gated on, it operates as a unidirectional (cardioid) microphone, keeping background noise at a minimum. Its design also makes it resistant to handling or clothing noise, vibration, and hum. Furthermore, the AMS28's wide (100

Hz to 10 kHz) frequency response has been specially tailored for high-quality speech applications, resulting in extreme clarity, without muddiness or boominess.

User net price of the AMS28 is $205.
SHURE BROTHERS INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 866-2553

NEW DYNAFEX INTEGRATED CIRCUIT FROM SSMT

Developed in conjunction with Solid State Micro Technology For Music, Inc., the new integrated circuit utilizing proprietary Dynafex noise-reduction technology is said to provide 30 dB of noise reduction on virtually any audio signal, without the encode/decode process. SSMT will manufacture and market the Dynafex SSM2200 chip under license.

Dynafex circuitry incorporates dynamically variable bandwidth limiting, along with downward expansion. Since these two types of noise reduction occur simultaneously, the designer claims a greater amount of noise reduction can be realized than in typical dynamic filtering schemes. According to MICMIX, the Dynafex circuitry provides a wider range of audio applications than a companding (encode/decode) system.

According to SSMT, production quantities of the new SSM220 will be available the first quarter of 1984. SSMT will be selling the chips to qualified OEMs who will be required to execute a sub-licensing agreement.
SOLID STATE MICRO
TECHNOLOGY FOR MUSIC, INC.
2076B WALSH AVENUE
SANTA CLARA, CA 95050
(408) 727-0917

UNITED MEDIA INTRODUCES ADR LOOPING CONTROLLER

The new Director and Translator units are microprocessor-based looping controllers for Automatic Dialog Replacement (ADR). Unlike the conventional synchronizer, these products utilize the same concepts as their time-proven counterparts in film production, by tying audio to video with full on-line editing capability.

The Director handles up to six "voices" separately, routing each performance to its own track. Standard functions include high speed "rock and roll" of tape and picture, storing loop input and output points, and recording each cue point as an automatic reassembly point for editing. The entire ADR can be pre-programmed using frame-by-frame accurate SMPTE timecode cueing with single keystroke control.

All data is stored on floppy disk for backup. The operator can, though the captive memory, scroll back through for cue review, or play through the tape. Both audible beeps and visual wipes are produced.

Standard interfaces accommodate one VTR and one audio transport as the minimum configuration, and expand to handle two more VTRs, or ATRs, or combination, based on the nature of work
UNITED MEDIA, INC.
4075 LEAVERTON COURT
ANAHEIM, CA 92807
(714) 630-8020

www.americanradiohistory.com

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www.americanradiohistory.com

news

medium, and slow versions). The necessary number of tracks to record one channel is one, two and four, for fast (30 IPS), medium (15 IPS), and slow (7½ IPS) versions respectively at a sampling frequency of 48 kHz.

Linear Packing Density is common to all versions, and specified at 1.51 kbits/mm (38.4 kbits per inch). Thanks to the newly developed modulation code, HDM-1, the minimum wavelength to be recorded (1.99 micrometers) is 50% longer than the conventional code, such as MFM. Double-track density (for example, 48 audio tracks on ½-inch tape at 30 IPS) is possible by using state-of-the-art thin-film heads, keeping the compatibility with the normal track density for the initial half number of tracks. In addition, cross fading can be provided in punching in/out; tape-splice editing; and electonic editing.

— News Notes —

Tannoy, Limited, the UK-based manufacturer of monitor loudspeakers and related products, has moved its head office to: Beadman Street, West Norwood, London SE27 0PW, England. (01) 670-1131. The company's products will continue to be manufactured at its factory in Coatbridge, Scotland... **E.A.R. Professional Audio** has relocated to a new 4,000-square-foot complex at: 2641 E. McDowell, Phoenix, AZ 85008. (602) 267-0600 . . . **Agfa Gevaert** Magnetic Tape Division has appointed Burlington Audio, Inc., 106 Mott Street, Oceanside, NY 11572, to market and distribute the company's audio tape products . . . From January 1, **Professional Audio Services** will be operating from the following address: 3710 West Magnolia, Burbank, CA 91505. The company's telephone number remains the same: (213) 843-6320 . . . **Ursa Major** has appointed the following five companies as domestic sales representatives: Givan-Flanagan Associates of West Boylston, MA: Lienau Associates, Inc. of Columbia, MA: Lassers, Sangwin, Lassers of Chicago, IL; R.L. Graham Associates of Leawood, KS; and Meyer, Ross & Fleming, Inc. of Burlingame, CA . . . **Crown International** has appointed Secom Systems of Chamblee, GA, and Teal Marketing of Atlanta, GA, to serve as sales representatives for the company's range of products . . . **MXR Innovations** has announced the opening of two repair centers to cover warranty servicing: Advanced Music Electronics, 2122-A S. Sepulveda, West Los Angeles, CA; and Music Dealer Service, 4700 West Fullerton, Chicago, IL. The new centers will honor all MXR warranties, regardless of the place of purchase. □□□

— People on the Move —

• **Ike Benoun** has been appointed president of Audio Industries Corporation's Hollywood-based operation. Benoun, recently president of Walt Davies Enterprises, rejoins AIC after an absence of two years, having been associated previously for 18 years. Appointed also to the board of directors, he succeeds **Hal Michael**, who becomes chairman of the board.

• **Thomas E. Mintner**, formerly broadcast products manager, has been promoted to the newly created position of director, Studer Products. In his new capacity, Mintner will carry responsibility for all sales and technical activities of the Studer Division for the U.S. market. Studer's New York City staff also has been augmented with the addition of **Nick Balsamo** as northeastern regional manager, and **Nancy M. Byers** as eastern regional sales engineer. Sales engineer **Vencil Wells** has been added to the staff of Studer's Southern California regional office in Van Nuys.

• **Juergen Wahl** has been appointed applications engineer for JBL and UREI products. In his new position at JBL, Wahl will provide technical support to the JBL/UREI sales and marketing organization, including product training sessions and consultations with JBL reps, dealer personnel, sound contractors, and other end-users. Also, **Debra Watson** has been appointed marketing services manager for JBL's Professional Division, and will be responsible for the creation and implementation of brochures and other promotional materials, and planning trade shows.

• **Gene Perry** has been appointed general manager of Audiotechniques, where he will direct an expansion program which will include enlarging the sales department, remodeling the New York headquarters, the establishment of a major MCI/Sony parts department, and construction of a digital audio editing and transfer suite.

• **Chuck Augustowski** has been appointed VP in charge of U.S. operations at Allen and Heath/Brenell USA of Orange, Connecticut. In addition to the new duties associated with his appointment as vice president, Augustowski will remain sales manager for the United States.

• **Paul A. McGuire** has been appointed vice president of marketing for Electo-Voice. Before this promotion, McGuire served as national sales manager.

• **Roger Miller** has been appointed western regional sales manager of Ampex Corporation's Audio-Video Systems Division, where he will direct the sales and service activities of the division's complete line of professional audio and videotape recorders. He succeeds **Tom Nielson** who recently was appointed national sales manager of the division.

• **Gerow D. (Gerry) Brill** has been named national sales manager for Clear-Com Intercom Systems, and will conduct a series of dealer/consultant seminars throughout the country, highlighting the company's new systems and interfaces.

• At Crown International, **Dr. Clay Barclay** will now serve as product development manager, and **Gerry Barclay** will manage the sales promotion activities. The Barclays joined Crown in 1981 to manage the company's sales promotion activities. Also, **Charles C. Hostetler** has been named as Crown's regional service manager for the northeast, and will supervise activities of service centers in his region and consult with them on service matters that require factory assistance. □□□

www.americanradiohistory.com

"No other microphone delivers hardcore rock 'n' roll like the SM57. That's why it's the only mic I use."
–Billy Squier

When Billy Squier rocks he needs a microphone that can roll with his punches. That's why he uses the Shure SM57. He's tried other microphones but he always comes back. As a matter of fact, the SM57 was the very first mic Billy used. It's taken him from small club dates to the big stages worldwide.

The SM57 is the perfect microphone for rock 'n' roll. It's got the right punch in live vocal applications to rise above the music. With a presence rise in mid-frequencies and a fixed low frequency roll-off to minimize "boominess" in close miking. The well-controlled polar pattern maximizes gain before feedback. Maximum gain is essential to high volume stage monitor applications during rock 'n' roll performances.

The SM57's presence peak makes it right for miking instru-

ments, too. With clean, well-defined accuracy.

The SM57 is a tough act to follow. Its rugged, reliable performance lets it stand up to the steamiest rock 'n' roll abuse. It can go on performing even when Billy Squier is ready to call it a night.

When it comes to hardcore rock 'n' roll the SM57 is a star. Just ask a star like Billy Squier.

For more information on the complete line of SM Microphones, call or write Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, (312) 866-2553.

SHURE®

THE SOUND OF THE PROFESSIONALS®...WORLDWIDE