RECORDING ENGINEER/PRODUCER

$3.50
December 1984
Volume 15 — Number 6

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

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December 1984
Volume 15 — Number 6

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RECORDING ENGINEER/PRODUCER

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

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- **Film**
- **Live Performance**
- **Video and Broadcast**

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— the Cover —

Why the mega-buck equipment investment in a "marginal" phonograph record production market? . . . a conversation with **MASTER SOUND's Bob Richardson** . . . exploring his decision-making process. (The questions asked by *R-e/p* that elicited the following statements should be fairly obvious.)

"To fully understand why we made the decision to buy SSL, it's important to understand the Atlanta market, our history in this market, and our objectives for the future.

"We came to Atlanta in 1964 to open a major music recording studio. We did our first session on June 4, 1964 — 'Hey Girl Don't Bother Me,' by The Tams for Lowery Productions and ABC Records. It was an instant hit and made the Top Ten. We had opened this facility with the best equipment available at the time: three-track, two-track

. . . continued on page 173 —

News Letters Views

OTARI MTR-12 REVIEW

from: **David Carlstrom**
service manager
ElectroMedia Service
Farmington Hills, MI

Peter Butt's detailed "Equipment Assessment" of the Otari MTR-12 [published in the October issue of *R-e/p*] is slightly flawed by inaccurate speculation on the cause or cure of a small amount of flutter.

The Otari's 9.95mm capstan diameter would yield a flutter component at 24.4 Hz, so the capstan is not responsible for the 27 Hz flutter Mr. Butt observed. 27 Hz more closely corresponds to the diameter of the inner race of the pinch-roller bearings. The small 0.035% to 0.063% flutter Mr. Butt measured seemed to me more likely to be a slight cogging in the MTR-12's pair of pinch roller bearings.

I checked the flutter of one new MTR-10-4 (four-track, half-inch). It read slightly lower than the machine Mr. Butt tested, but did have the 27 Hz component: it measured 0.02% flutter while recording, and 0.04% flutter after rewinding and playing.

Reducing the pinch-roller force from 2.5 kg to 1.9 kg removed the 27 Hz component from the flutter residual, confirming the pinch-roller bearings were responsible for this flutter. This MTR-10-4 now reads at most 0.02% on a tape recorded, rewound, and played.

Otari recommends fine tuning pinch-roller force for optimum starting in the range of 2.5 kg to 1.7 kg. The only possible harm of optimizing pinch-roller force for low flutter might be a less than optimal start, which may not be important to users requiring minimal flutter. Those interested in obtaining the lowest possible flutter from an Otari MTR-10-12 should experiment with reduced pinch-roller force and possibly selection of pinch roller bearings.

Reply from: Barry Ross
manager of field service/
engineering, Otari Corporation

In the three years of MTR-10 history, and the recent tests we performed on the MTR-12 (using similar wow and flutter test equipment to Peter Butt and Dave Carlstrom) we concur that Dave Carlstrom's findings regarding the location of the 27-Hz flutter component and adjustment cure are most accurate. Although, as Dave mentioned, by adjusting the pinch-roller pressure to a value just below 2.5 kg, the machine will respond with a slightly slower start-up time. The slower start is more apparent, of course, when using half-inch tape due to the increased weight, which we feel is a factor just as important to some Otari users as wow and flutter. Otari does support the adjustment of the pinch-roller pressure to suit the customer's needs, although we ship all MTR-10s and MTR-12s set up with 2.3 to 2.5 kg of pinch-roller pressure, in an attempt to split the difference and please all users. It should be noted that in our service manual, which accompanies each machine, there is no mention of this fine pressure adjustment. To simplify the procedure, the manual just clearly states that the user should set the roller pressure to 2.5 kg.

We invite our MTR-10 and MTR-12 users to call on us if they require any assistance in the above-mentioned pressure adjustment. □□□

NB: An additional reply from Peter Butt can be found on page 178 — *Editor.*

STUDER HEADS

from: **Thomas E. Mintner**
VP and general manager
Studer Revox America, Inc.
Nashville, TN

While reading through the October issue of *R-e/p*, we have noticed an advertisement which requires clarification. The Sprague Magnetics advertisement on page 32 asks the question "Do you know who manufactures the audio heads for your recorder manufacturer?" Apparently Sprague does *not*, as they attribute our Studer audio heads to Woelke of Germany.

Willi Studer AG manufactures its own audio heads in our main factory in Regensdorf, Switzerland. Woelke is certainly a fine company, and in fact we do utilize them for our *erase* heads. However, our entire audio head manufacturing department in Switzerland would be quite disturbed to learn that representations (apparently accidental) were being made that the heads for which we feel Studer is justly famous were made other than by Studer.

Thank you for the opportunity to set the record straight. □□□

"Letter to the Editor" from Welton H. Jetton and Jeff Paullus, October Issue — A Correction

A serious typographical error in a "Letter to the Editor" published on page 16 of the October issue of *R-e/p* may have resulted in a misleading parameter being assigned to the frequency-response capability of Auditronics' products. In the additional comments section by Jeff Paullus to a reply by Welton H. Jetton, president of Auditronics, Inc., we inadvertently referred to the fact that a "flat low-frequency response to 5 *kHz* (-1 dB) would suffice in the real world." This should, of course, have read: "flat low-frequency response to 5 *Hz* (-1 dB) would suffice in the real world."

Our apologies to Welton Jetton and Jeff Paullus for any embarrassment that may have been caused by the error — *Editor.*

WEATHER PROTECTION

from: **Y. Brevda, president**
Yale Audio of Florida
Tampa, FL

I read with interest David Scheirman's article "Anatomy of an Injunction" in the August issue of *R-e/p*.

Per the lawn area speakers, I note the mention that their amplifiers and crossovers are housed in weatherproof cases. What is in the design of this system to protect the speaker enclosures and their drivers from adverse weather? And what is done to protect the main stage area speakers from adverse weather?

And how are *both* sets of speakers protected from damage during a show when it starts raining? From the photographs published in the article, the stage speakers at both sides appear to be located out in the open with no overhead coverage.

Also, the article mentions that there are seven lawn speaker towers. What kind of a problem does this pose for any viewing obstruction?

David Scheirman replies:

As this system is used only during the summer months at a seasonal performance venue in relatively dry Southern California, protection from adverse weather is not nearly such a critical concern as it might be in other parts of the country. For a good explanation of precautionary measures taken with a permanently installed system that operates in a harsher climate, refer to Paul D. Lehrman's excellent article, "High-Quality Sound System Design for New England's Sullivan Stadium," published in the April 1984 issue of *R-e/p*.

Michael Adams of Sound Image, contractor for the Pacific Amphi theatre's new sound system, replies:

All of the speaker cones were treated with Scotchgard™ at the beginning of the season. The main speaker stacks are shielded by nylon-mesh black scrim. Those loudspeaker units atop the delay-line towers are faced with a weather-resistant grillcloth. We did have one or two cloudbursts, but in general the weather was so hot and dry for this past season that rain damage was not a problem.

As for the viewing obstruction problem posed by the lawn-speaker towers, we

. . . *continued on page 11* —

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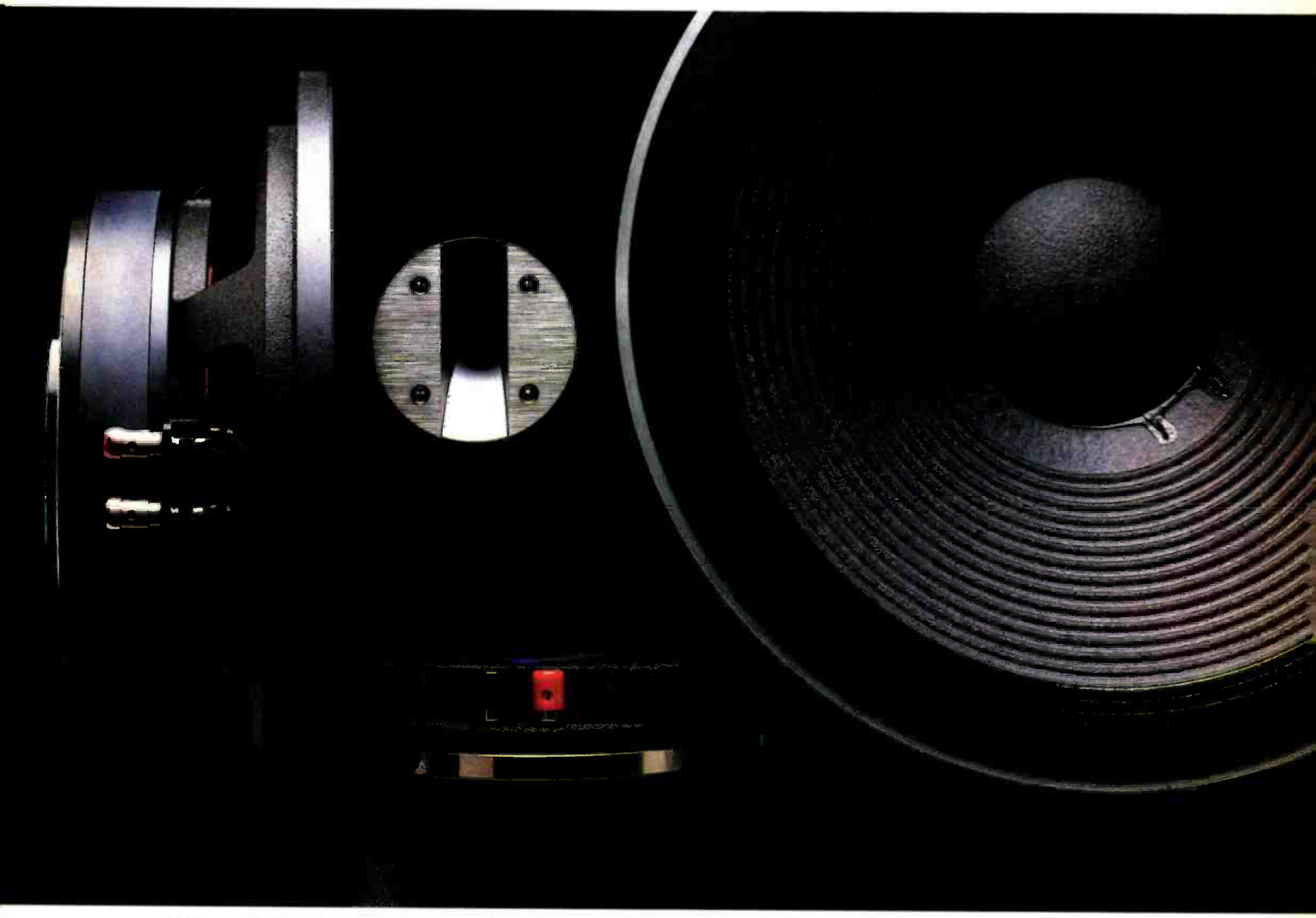

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For additional information circle #6

Letters

— continued from page 6 . . .

found that the only way to get through the crowd during capacity shows was to walk up the narrow aisleways left on the sloping lawn behind the poles. It was certainly not set up that way by design, but, since there are no "skyhooks" to suspend the speakers from, those open lanes made it quick and convenient to get to the rear of the seating area to check sound levels and coverage. ☐☐☐

ALPHA STUDIOS AND BTX SOFTOUCH

from: **Paul G. Matthews, product manager**
The BTX Corporation
Bedford, MA

Ralph Jones' article on Alpha Studios in the October issue was as comprehensive a piece of reporting as I've yet seen on one studio's response to the demand for high-quality audio-for-video and film. The sidebar on our Softouch system is a good, in-depth look at its versatility and power in many types of audio editing situations. Kudos to *R-e/p* and Mr. Jones.

The inclusion of several Softkey command sequences used by [Alpha owner and chief engineer] Gary Brandt was innovative on your part, and one we were understandably delighted to see. It not only provides other Softouch owners with useful Softkeys developed by a knowledgeable user, but also powerfully illustrates exactly what a programmable Softkey is to those who are not familiar with the Softouch Audio Editing System.

In addition to praising your article, I'm compelled to point out a few errors in the command sequence as printed.

- In Program #2, the "minus" key must be pressed after TRIM, so that the 10 seconds are *subtracted* from the cue point. Also, the ENABLE RECORD (after the MARK IN command) should be moved to replace one of the SLAVE ASSIGNS entered after LOOP BEGIN. Otherwise, the loop would not be a Preview, as Gary intended, but a real record.
- Gary's Softkeys are good examples of consistency within the Softkeys. In each case, he uses the Softkey to set up a loop, then Preview it, then get the RECORD and ASSIGNs set for the "live" take. In keeping with this structure, the SLAVE ASSIGNs at the start of Programs #1 and #3 should not be included.

Anyone who would like a complete copy of those Softkeys with the changes and other streamlining included, can contact me at the BTX Corporation, 75 Wiggins Avenue, Bedford, MA 01730.

We are proud that our Softouch system, in Gary Brandt's skillfull hands, is playing a part in the success of Alpha Studios. Thanks to *R-e/p* for documenting the contribution of an audio editing system to a studio's growth. ☐☐☐

LEAVE IT ON?

from: **Brian C. Carr, studio manager**
Walk On Water Studios, Inc.
New Braunfels, TX

I would like to ask a question that always rears its ugly head, yet no one I know can agree on what is safe for the equipment and technically acceptable. Each time I install some studio gear, the guy I put it in for and I have a nice debate — but no mutually acceptable solution is agreed upon — so perhaps if you could pass the question to the people in the know. They can end this debate once and for all.

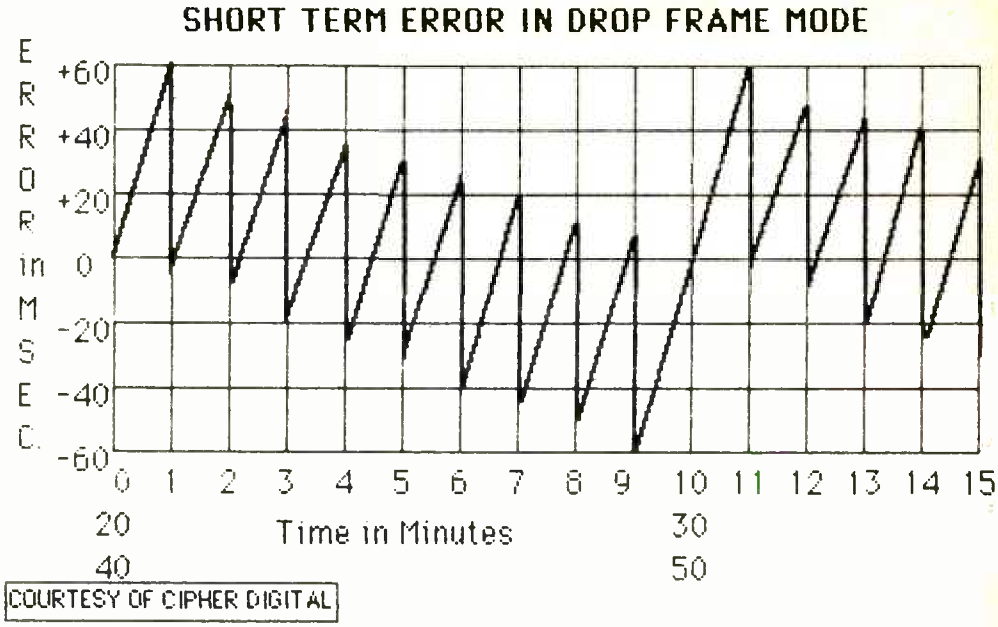
Should all solid-state electronic devices be left powered up constantly to maintain its temperature, and therefore its thermo-electrical properties, or can we subject the unit in question to the normal surges and discharges made daily during power-up and power-down and expect it to work like a champ with no degradation to the normal life? The question pertains mostly to digital reverbs, consoles, DDLs, power amps and the usual modern outboard gear in

Letters

daily use. Of course, capstan motors, incandescent bulbs and fan motors have a shorter life span than the above, and should be turned off where possible (e.g., leaving the half-track tape sensor disengaged when not in use) and are disregarded in this question.

Secondly, my reaction to the article on 12 different microphone "sound" characteristics ("Performance Assessments of Twelve Studio Microphones", published in the April 1984 issue of *R-e/p*) is that I personally believe the test to be a valid one. When miking vocalists I am not familiar with I rely on the "press conference" technique used by Professor Cross. Remember: Every studio and control room has its own characteristics working against the anechoic specs.

Editor's Note: John Roberts addresses the question of continuous powering of studio equipment in his regular column "Exposing Audio Mythology." Also, a second article by Professor Cross can be found elsewhere in this issue. □□□



TIMECODE CONFUSION

from: **Steve Krampf**
VP sales and marketing
Otari Corporation
Belmont, CA

"Is there any difference in the frame rate for *D*rop *F*rame and *N*on-*D*rop *F*rame Longitudinal SMPTE timecode in NTSC color video?"

I have informally posed this question recently to a good number of people from the audio industry now working in Audio Post Production. An alarmingly high percentage of some very knowledgeable people have answered that DF and NDF timecode had a different frame rate, *i.e., speed.* They responded that DF code had a frame rate of 29.97 *f*rames *p*er *s*econd (actually 29.97002617*) while NDF code had a frame rate of 30 fps.

Additionally, many of these same people responded that it really wasn't that important what the code frame rate was because the speed differences could be compensated for at a later date.

It is our firm contention that standard engineering practice calls for both DF and NDF to be the *same* frame rate, and referenced to vertical sync (59.94 Hz and multiples) for NTSC color video and synchronized sound. Unfortunately, studios doing APP must buy an NTSC sync or color bar generator.

For further reference, I would highly recommend that the *R-e/p* reader obtain a copy of "The Time Code Handbook" written by Walter Hickman, and published by Cipher Digital, 150 Huntington Ave., Boston, MA 02115. Although it is temporarily out of print, I have been told that Walter Hickman is expanding the book and it will be available in six months. Another source of information is the EECO Corporation's handbook, available from EECO, 1601 E. Chestnut Avenue, Santa Ana, CA 92701. For further information about why NTSC uses 59.94 Hz instead of 60 Hz, we recommend the Ampex Video Training Manual, available from Ampex Corporation, 401 Broadway, Redwood City, CA 94603.

This letter covers only one area of APP, and begs the larger question of mixed time bases, *i.e*,, film rates and conversions. I encourage the media to conduct a forum on practices with regard to all aspects of *A*udio for *F*ilm and *V*ideo.

Some Finer Points:

"SMPTE timecode is *not* timecode (ref: to 59.94 Hz); EBU timecode *is* timecode."

NDF SMPTE timecode generates an error of 0.03 frames per second between its time count and the "clock on the wall," which amounts to approximately a 60 ms error per minute. DF SMPTE code compensates for these "timing" errors according to the accompanying error chart.

As can be seen, DF timecode is only "timecode" for an instant, every tenth minute.

We therefore suggest that we begin calling SMPTE timecode **SMPTE Edit Code.**

In the case of 50 Hz-based EBU timecode, none of the above problems exist. Referencing your timecode to line frequency, PAL vertical sync (all 50 Hz) are exact intervals of the frame rate, 25 fps; there is no drop or non-drop frame code. EBU code is *absolute* timecode.
□□□

* The difference between 29.97 fps and 29.97002617 fps introduces an accumulating error of +75.442 milliseconds per 24 hours. This error is judged to be within the timing requirements of most broadcast and production plants. It is suggested, however, that this "long term" timing error be corrected approximately once a month — *SK*. □□□

Voice Design for "Gremlins" — A Correction

Mark Mangini and Larry Blake would like to correct an error that crept into their article on the voice design of *Gremlins*, published in the August issue of *R-e/p*. In the article, they credited Terry Eckton with the editing of Darth Vader's voice in the *Star Wars* films. Instead, Vader was cut by Bonnie Koehler; Ms. Eckton was responsible for Chewbacca the Wookiee. As fate would have it, Ms. Koehler supervised the sound editing of the foreign versions of *Gremlins*.

SUBJECTIVE LOUDNESS

from: Thomas D. Rossing
Professor of Physics
Northern Illinois University
DeKalb, IL

In the August 1984 issue is a long letter from Rick Simon, part of an exchange of letters with Alan Fierstein. I shall avoid entering their discussion about power potentiometers, but I would like to correct a couple of errors about the scale for subjective loudness.

So far as I know, S.S. Stevens proposed the first *sone* scale in 1936 (*Psychol. Rev., 43,* 405-16). In a later paper (*J. Acoustical Soc. Am., 27,* 815-829 [1955]), he tabulated the values various observers had reported for a 2:1 loudness ratio. These values ranged all the way from 2 dB to 24 dB, and Stevens suggested using 10 dB as the best value. This of course leads one to the expression recommended by the International Standards Organization for relating loudness S in sones to loudness level L_s

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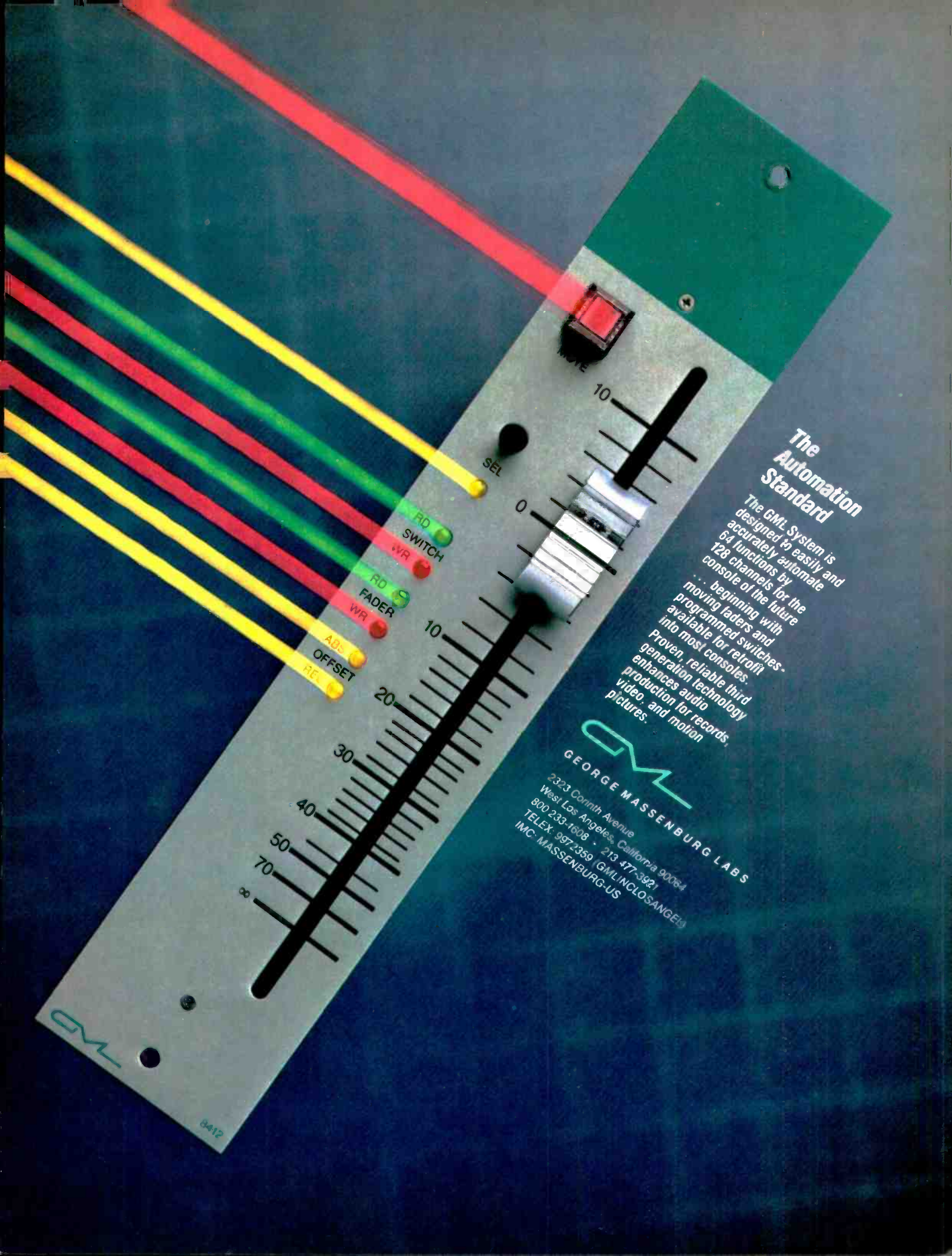

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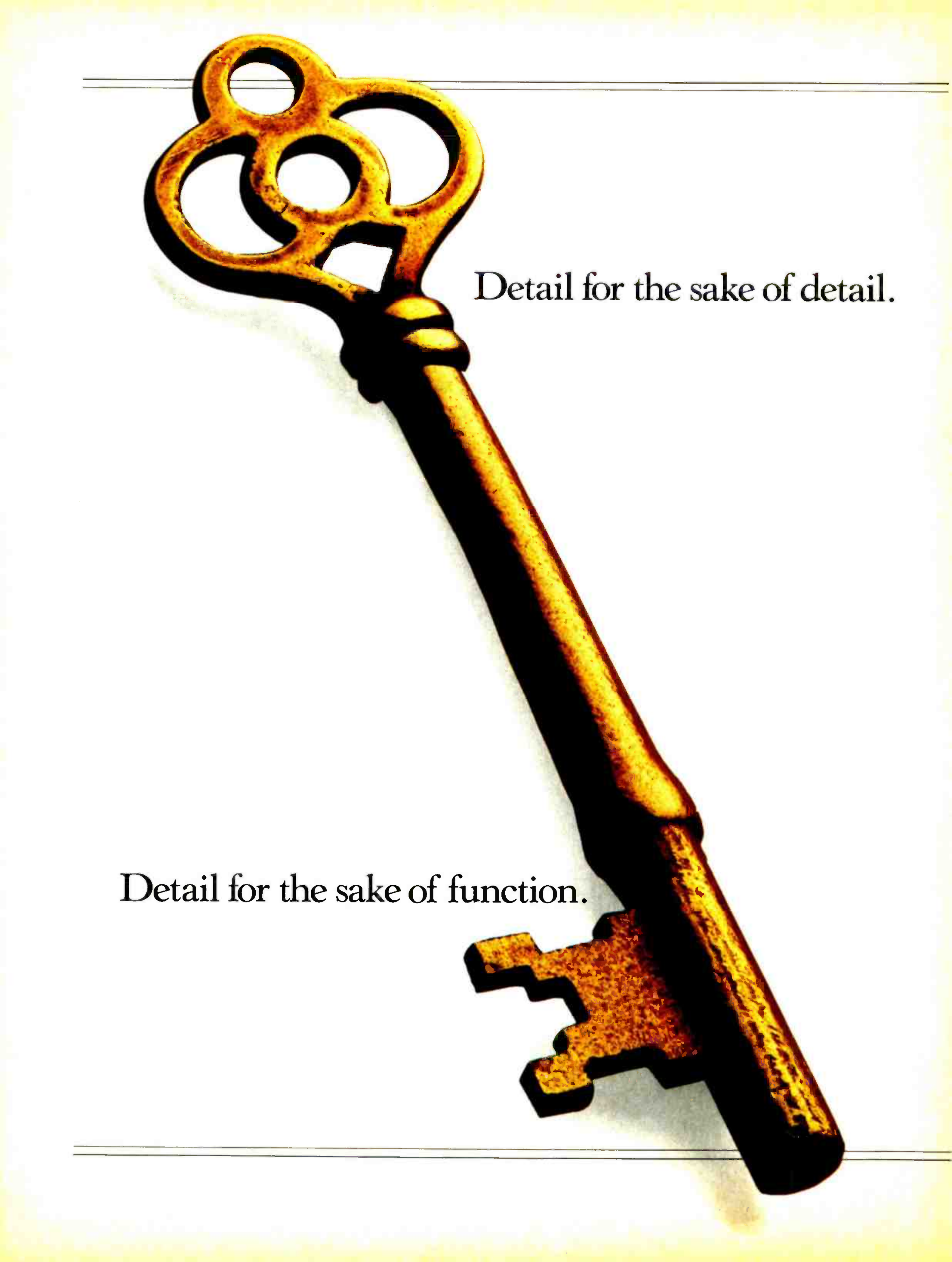
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- Three color display for the best response to peripheral vision
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- One quarter dB resolution at "O Vu" for accurate calibration and tape machine alignment
- Variable intensity function allows LEDs to be adjusted for optimum brightness in a wide range of control room lighting conditions
- Individual calibration for Peak and Average modes
- High-slew balanced input circuit prevents ground loops and high frequency inaccuracies
- Modular design for easy maintenance access

In addition to metering console and tape machine levels, each Series 34 I/O module also contains an additional input meter to monitor the input channel signal path post the equalizer. This additional meter helps the engineer optimize gain staging throughout the console, yielding maximum headroom and optimum signal/noise.

Some may say that our attention to detail has produced a metering section that seems *over-designed*. Perhaps, but from a functional standpoint, it will help you make better sounding recordings.

Equalization: Although we pioneered the state-variable design used almost exclusively in today's modern parametric console Eqs, the Series 34 uses a simpler design which is not fully parametric. Why? More attention to detail.

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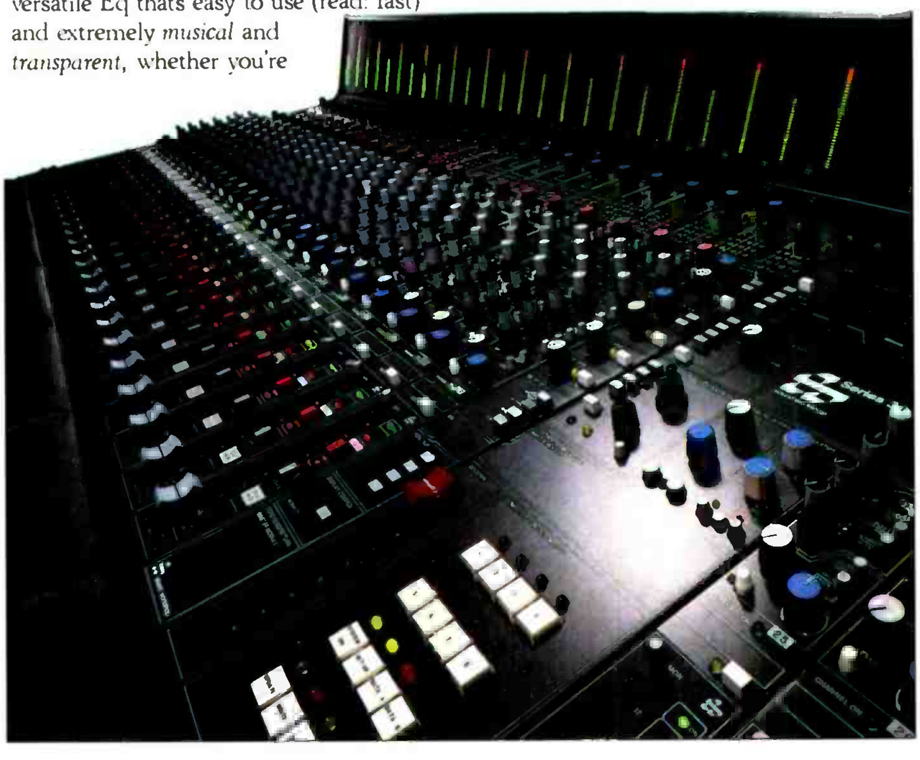
Simply put, the Series 34 four-band sweepable Eq sounds better than any console parametric equalizer we know of. Its wide frequency overlap between bands, low active stage count and intelligent PC layout has yielded an extremely versatile Eq that's easy to use (read: fast) and extremely *musical* and *transparent*, whether you're altering the frequency balance just a pinch or at its plus/minus 15 dB maximum. Our Series 34 Eq won't notch out a guitar amp buzz—we'll let one of your rack Eqs do that for you. It will simply allow controlled tonal balance changes, whether subtle or extreme, with complete sonic integrity.

Some may say that our attention to detail has produced an equalization section that is *under-designed*. Maybe, but functionally, you'll be able to make better recordings.

Sound Workshop's metering and equalization are just two examples of our relentless attention to detail throughout the design of the Series 34. Its high performance and extremely wide dynamic range (output clip level exceeds +27 dB) are enhanced by each section of the console whether it's sonically passive, as in the metering, or sonically active, as in the equalization. Besides giving you that extra edge you need at your next session, the Series 34 is designed to make sense.

The Sound Workshop Series 34... It makes sense.



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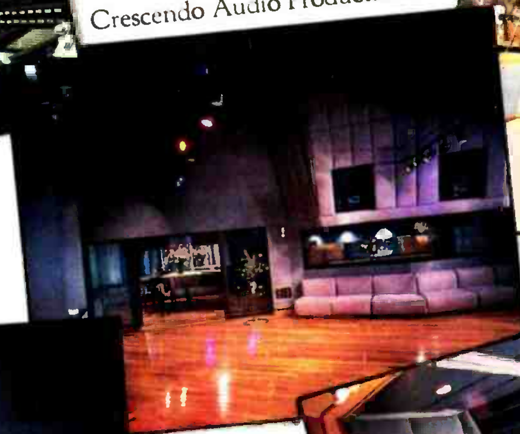

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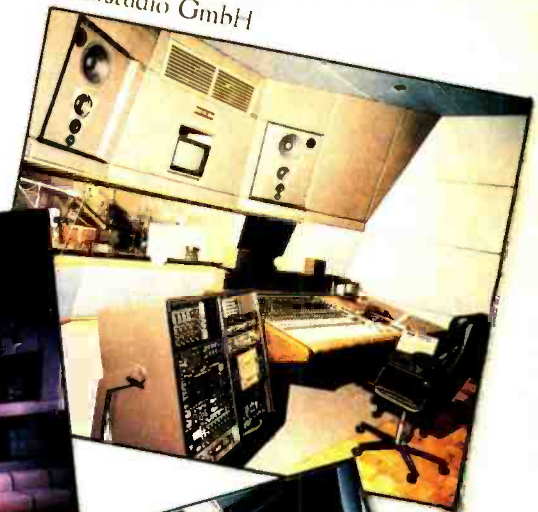

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For additional information circle #10

— continued from page 12 . . .

Letters

in phons:

$$S = 2 \times [(L_s - 40)/10]$$

from which it can easily be shown that $\log S = 0.0301\,(L_s - 40)$, slightly different from the expression in Simon's letter. An equivalent expression for S that avoids the use of L_s is:

$$S = C \times p^{0.6}$$

where p is the sound pressure and C depends on the frequency. This expression may be preferred, because sound pressure is what we measure with a microphone or sound level meter.

Several investigators, however, have found a doubling of loudness for a 6 dB increase in sound pressure level (see, for example, *J. Acoustical Soc. Am., 48*, 1397-1403 [1970]), rather than a 10 dB increase. This suggests a formula in which loudness is proportional to sound pressure (see *Acustica, 30*, 247 [1974]):

$$S = K \times (p - p_0)$$

All of this is for tones of a single frequency. Predicting the loudness of complex sounds (such as music) is more difficult. May I refer the interested reader to my book *The Science of Sound* (Addison-Wesley, 1982), pages 86-89? Two well-known methods for predicting the loudness of complex tones are known as "Stevens' method" and "Zwicker's method". Table 6.3 in my book, however, suggests a simple relationship for estimating loudness (sones) from A-weighted sound levels. It is not exact, but it is useful.

The 1937 paper by Harvey Fletcher (not J. Fletcher) and his colleague W.A. Munson (*J. Acoustical Soc. Am., 9*, 1-10) uses a loudness number N rather than a scale of sones. For loudness levels above 40, N appears to increase by a factor of 10 when the loudness level increases 30 dB. This can be expressed:

$$N = 10^3 \times 10\,[(L_s - 40)/30]$$

from which $\log N = 3 + 0.033\,(L_s - 40)$. But this Fletcher-Munson loudness number N is almost never used nowadays.

The sentence in Simon's letter which begins "what all this means is . . ." is quite garbled and should probably be ignored altogether, by the way.

Rick Simon, president of Simon Systems, replies:

The sentence in my August letter "What all this means . . ." is unclear and I had planned to write a supplement for it. It was not my original intention to go into a lengthy discussion about subjective loudness (that letter was long enough), but Professor Rossing brings up certain issues which bear qualification, especially as to how they pertain to the original subject of volume control, and more specifically volume control for headphone distribution systems.

One available reference for the equation $\log L = 0.033\,(LL-40)$ is Kinsler-Frey *Fundamentals of Acoustics*, Second Edition, (John Wiley & Sons; New York, 1962). I would like to refer the interested reader to pages 392 to 398. Figure 13.12 plots Subjective Loudness (L in sones) versus Loudness Level (LL in phons) for a pure tone. The above equation is an approximation of the graph over the range from 40 to 100 phons (comfortably audible to unpleasantly loud). From this approximation it can be easily shown that:

$$L = C\,(I)^{1/3} = C'\,(\Delta P)^{2/3}$$

where I = Intensity, P = Pressure, C and C' are frequency dependent parameters.

These equations, which are slightly different from Rossing's letter, can also be found in *Introduction to the Physics and Psychophysics of Music*, by Professor Juan. G. Roederer (Springer-Verlag; New York, 1973), and Stevens (1970). They indicate that if loudness is proportional to the cubed root of Intensity, doubling the subjective loudness requires that the power supplied to the sound source increase by a factor of eight; or, stated another way ⅛ the power for half loudness. The expression that uses intensity may be preferred because intensity is directly related to power, which can be supplied to a set of headphones under the control of a potentiometer. More recent studies, however, have suggested that the second Stevens Sone Scale and power-law relationship which dominated acoustical thinking for years, was not bias free.

The 1970 paper in *J. Acoustical Soc. Am., 48*, by Richard Warren, that Professor Rossing refers to, involves subjective testing and attempts to eliminate experimental biasing. This paper concludes that for judgements of 50% loudness, L is proportional to the square root of intensity. Since pressure is proportional to the square root of intensity, this suggests a relationship where loudness is proportional to sound pressure. Figure 3 in the paper shows that 50% loudness judgements correspond to 6 dB attentuations over the range of 45 to 92 dB intensity levels. It also shows, however, that in the ranges above and below the 45 to 92 dB range this linear relation no longer exists. To quote Warren directly: ". . . Figure 3 shows that for the 100 dB St (St = standard intensity) a 94 dB Co (Co = comparison intensity) is significantly greater than half as loud, and that for the 36 dB St a 30 dB Co is significantly less than half as loud." If we are going to talk about volume control, we need to define these outer ranges, especially in the lower range.

Walton Howes, in his 1974 *Acustica* paper (Vol. 30 pp. 247 to 259), approaches the issue of loudness from physical rather than psycho-acoustic quantities. He derives a loudness function based on electrical discharge rates in auditory nerve fibers, rather than human judgements. He summarizes the loudness function by dividing it into four subranges. Figure 5 in his paper plots a curve of Loudness versus Sound Pressure for a 1 kHz tone. The four subrange equations which describe this curve are:

1. Dynamically active subrange
   $$(0 \leqslant S \leqslant 34\text{ dB})\quad L = 10^{2}\,(\ln q + 12.4)\,(q - 2 \times 10^{-5})$$
2. Active Subrange
   $$(34\text{ dB} \leqslant S \leqslant 90\text{ dB})\quad L = 5 \times 10^{2} q$$
3. Fiber Saturation Range
   $$(90\text{ dB} \leqslant S \leqslant 120\text{ dB})\quad L = 10^{2}\,(4.4 - \ln q)\,q$$
4. Fiber and rate saturation subrange
   $$(120\text{ dB} \leqslant S)\quad L = 3000$$

where S = sound pressure level, and q = effective sound pressure.

The issue here is volume control, and that requires a method for adjusting levels from zero to some maximum value, not just cutting loudness in half. I originally brought up the issue of subjective loudness in my August letter, because of Alan Fierstein's recommendation to use linear instead of audio tapered pots for controlling volume in headphone distribution boxes. Power dissipation was his main concern. My August letter fully covers this topic and proves that a properly designed system will not be subject to power dissipation problems. This leaves only the question of which type of taper will help the listener to perceive a more linear amplitude response as a function of percentage pot rotation.

A linear tapered pot has essentially one range and one slope. While this type of taper might suffice for subrange 2 of Howes loudness curve, it would become deficient in subrange 1. In keeping with Warren's findings at low levels, this could explain why using a linear tapered pot as a volume control causes such a drastic level change in the lower region of percentage rotation. The audio tapered pot (such as the type I previously recommended), is divided into two major subranges: the first subrange from 0 to 60% rotation is a somewhat exponential (e to the power of the square root of percentage rotation); in the second subrange from 60% to 85% rotation the pot becomes linear.

I strongly recommend the use of audio tapered pots over linear pots in volume control applications such as the aforementioned. I would suggest that anyone doubting this recommendation perform the simple test of replacing an audio tapered volume control with the equivalent resistance linear pot, and compare results. Those of us who have designed and worked with audio equipment are well aware that the subjective loudness appears to change more linearly with percentage pot rotation when audio tapered pots are used.

Finally, the principles used by Howes can also be applied to predicting the loudness of a complex sound. If you accept this newer theory based on physical quantities, may I suggest you read his 1979 *Acustica* paper (Vol. 41, #5, 1979, pp. 277 to 320) on this subject rather than Rossing's book. The Stevens Method and Table 6.3 in Rossing's *The Science of Sound* are based on the older psycho-acoustical data (10 dB increase for a doubling of loudness) which Rossing in his letter implies are not valid. □□□

EXPOSING AUDIO MYTHOLOGY

Laying to Rest some of the Pro-Audio Industry's more obvious "Old Wives Tales"

by John Roberts

The topic for this month's column is impedance. While most of us are familiar with the definition "Impedance equals resistance plus reactance" (reactance being the part of a component's resistance that changes with frequency), I would like to focus on how impedance considerations relate to interfacing equipment. In Part Two I will attempt to shed some light on the question of leaving equipment on, or turning it off when not in use.

Matching Impedance

A popular misconception when interfacing audio gear is that the input impedance of one stage should equal the ouput impedance of the preceding. In fact there are only two cases where it is beneficial, so let's take a look at both.

A. Maximum Power Transfer

It is convenient, when analyzing circuits, to model practical signal sources as perfect voltage sources (zero-ohm output impedance) with a fixed series resistance (Figure 1). This technique reliably predicts the voltage drop caused by internal losses as more current is delivered by such a signal source to a decreasing load resistance.

You will note from Figure 2 that maximum power is delivered to the load R_l, and thus maximum power transfer occurs for the unique case $R_l = R_s$. As profound as this may seem, it holds limited utility for the modern circuit designer. For example, connecting your 10-ohm source impedance moving-coil cartridge directly to your 8-ohm speaker system will enjoy near-optimum power transfer, but won't get very loud!

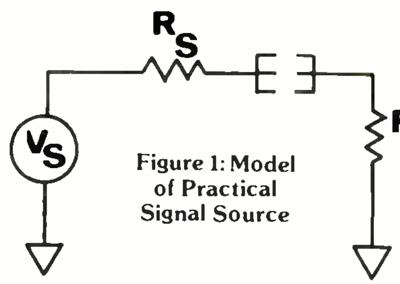

Figure 1: Model of Practical Signal Source

Perhaps a better example in which maximizing power transfer does have some merit is the case of a vacuum-tube power amplifier. An output tube with a 1 kohm plate resistance would dissipate 62.5W for every 1W delivered to a 16-ohm speaker (8 ohms is even worse). An 8:1 voltage step-down transformer would cause a 64:1 transform in reflected impedance — since impedance varies as the square of the turns ratio — for a good match between speaker load and tube. The 16-ohm load will now receive 16W while the tube dissipates 16W (for the same voltage), in this case maximizing the power output for that particular tube.

Modern power amps, thanks to the low output impedance of semiconductors and benefits of negative feedback, operate with output impedances often less than 0.1 ohm, catering to more practical concerns such as efficiency, damping, and frequency response.

B. Terminating transmission lines

The other case for matching impedance is when interfacing with transmission lines. As you may recall from my discussion of speaker cables in the October 1983 issue, a simple wire pair will have reactive as well as resistive components. Such a cable will have a characteristic impedance. When termin-

. . . continued on page 23 —

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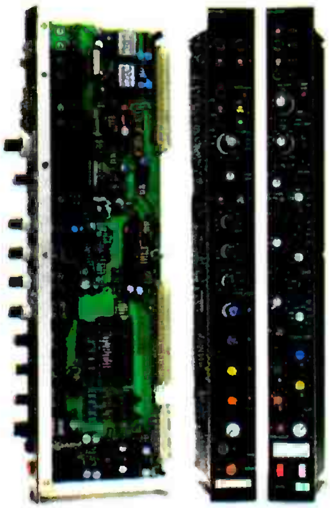

Basic input modules feature 3 or 4 band EQ, microphone/line inputs, 5 pre/post-fade auxiliary sends, and channel overload indicators. Options include transformerless mic preamps on a subcard, separate transformerless TAPE input for remix, stereo input modules, stereo EQ, internal stereo X-Y/MS active matrix, stereo blend control, dual line inputs, variable HP and LP filters, user defined panel switches, and the list goes on.

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For more information on Studer consoles, call or write: Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

STUDER REVOX

For additional information circle #13

— continued from page 18 ...

ating such a cable with something other than its characteristic impedance, reflections can cause part of the energy to bounce back up the line, to produce interference and incomplete signal transfer.

To better visualize this phenomenon, imagine two boys holding opposite ends of a long rope. The first boy quickly whips his end causing a pulse to travel down the rope. When it reaches the second boy it reverses and travels back up the rope. The second boy holding the rope rigid is equivalent to a short circuit at the end of a transmission line.

Let's now visualize that the first boy is also part magician, and can hold the rope up by himself. If he again sends a pulse down the rope, when it reaches the free end the pulse will again reverse itself and return back up the rope (where else could it go?). This is equivalent to an open or bridging circuit termination.

Now suppose that boy #2 is also part magician and conjures up and infinitely long rope which he connects to his end. When boy #1 sends the next pulse it disappears down the infinite rope, never to return. This is equivalent to a properly terminated line. (Our infinitely long rope could have been replaced by a specifically damped spring with the same characteristic impedance for an identical result.)

Most electrical transmission lines are terminated with a resistance or a resistance and a suitable reactance, to properly model the characteristic impedance of the line being terminated. The sensitivity of a line to improper termination is a function of how long the line is; how short the wavelength of the signal you are sending; and the propagation speed of the signal.

Audio signals are relatively low frequency to be bothered by this effect, but very long audio lines, such as telephone cables, are *always* terminated. Video and computer networks will usually be carefully terminated; proper termination is also very important at radio frequencies. (You wouldn't want to send 50,000 watts to your radio station antenna, only to have 25,000 reflect back to you!)

While I cannot quote a magic number of feet for which you must begin treating your audio wiring line as a transmission line, if you are sending a signal across the room don't worry about it; if you are sending it across town . . . worry! (Note: telephone lines want to be driven by 600-ohm source impedances and expect 600-ohm terminations. Video and high-speed computer lines are designed around 50- or 75-ohm impedances. Good old two-conductor shielded will typically look like 50 ohms should you even encounter a few thousand feet of it in one piece.)

Interfacing Equipment at Audio Frequencies

The most common "short haul" audio interface is what's known as a bridging

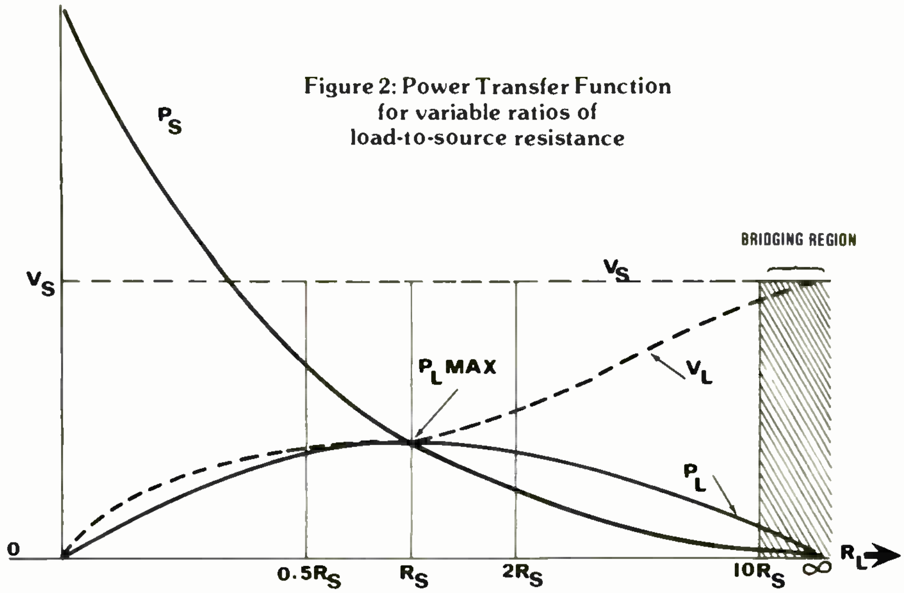


Figure 2: Power Transfer Function for variable ratios of load-to-source resistance

connection. The output impedance of the preceding stage will be several times lower than the input impedance of the following stage. The intention is to maximize voltage transfer with no consideration of power transfer, with the exception of power amplifiers where the intention is to increase efficiency by wasting as little power as possible in the amplifier's output impedance.

Modern electronic devices are strictly voltage-in/voltage-out devices, and care must be taken when trying to evaluate them by classical yardsticks. Source impedance and output impedance are *not* always equivalent. Output impedance of active circuitry relies upon negative feedback and linear operation of the internal electronics to remain predictable. Operation outside of design voltage or current parameters will result in non-ideal performance.

For example, a power amp with a 0.1 ohm output impedance will not deliver maximum power into a 0.1 ohm output speaker, not even until it blows up! Likewise, a line-level processor with a 10-ohm output impedance will not be very happy driving a 10-ohm load.

Output impedance is a potentially confusing specification; it is more important to know how low of an impedance the device was designed to drive. Most line-level processing gear falls into two categories: those that will drive 600-ohm loads, and those that won't. Since a general purpose op-amp will only drive about 2,000 ohms, most low-cost (semi-pro?) and almost all hi-fi gear will not appreciate low impedance terminations. For that reason most modern equipment is designed with at least 10,000 ohms input impedance; when a lower input impedance is required it's simple to add a resistive shunt.

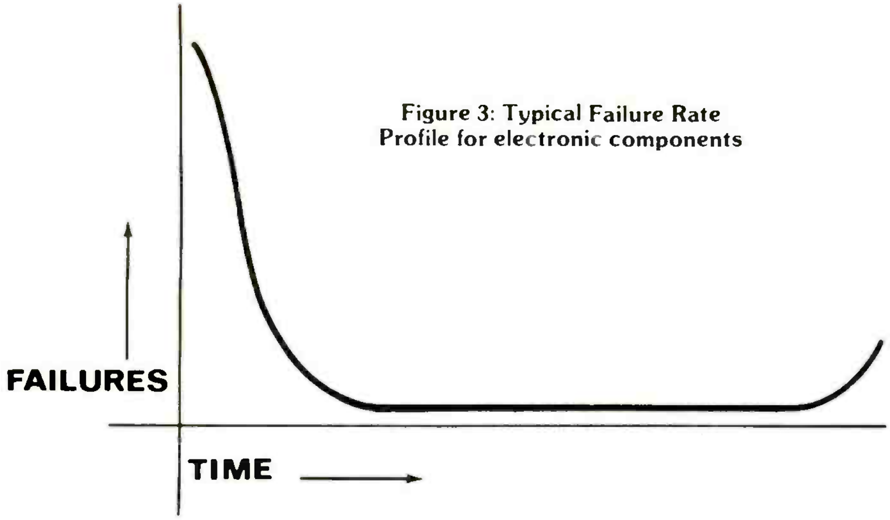
Generally speaking, impedance can be all but ignored when interfacing line-level gear. However, there is another exception worth discussing. Signal distribution systems that use transformers to balance and isolate lines are commonly operated at a nominal 600 ohms impedance. There are several advantages to running such systems at low impedance, including wider bandwidth, lower distortion, less crosstalk/pick-up, etc. I don't expect transmission line considerations to be a significant factor. The popularity of 600 ohms is a testament to the influence of telephone system engineering on early audio system designs, and the fact that a lot of broadcast audio still gets connected to telephone company land lines.

TURN OFF OR NOT TURN OFF, THAT IS THE QUESTION!

First, I would like to thank Brian Carr, studio manager of Walk on Water Studio, New Braunfels, Texas, for his excellent question ["Letters to the Editor," page 12], and encourage other *R-e/p* readers to put pen to paper if they have a similar question that has evaded solution. I may not know the answer either, but will try to find out.

The question of whether it is better to leave your studio powered up continuously does not offer any simple answers. In the days of vacuum-tube circuitry, the benefit analysis was pretty straightforward. Not unlike the incandescent light bulb, there is a roughly linear deterioration versus on-time wth an incremental mechanical stress caused by each thermal on/off cycle. For any off period where the linear deterioration would exceed the damage of one on/off cycle, it is desirable to turn the unit off. While this breakeven off-time will vary from unit to unit, I feel confident suggesting that tube equipment be powered down overnight.

It is not so simple in the case of solid-state electronics. The linear deterioration mechanisms are much more gradual, and more likely to be measured in years than hours. An integrated circuit, when operated within its specifications, will probably last a lot longer than our desire for the product around it. In the real world, however, semiconductors



don't last forever. Since they usually don't wear out, an analysis of whey they fail might shed some light on the powering down question.

For the sake of this discussion, I will assume that the semiconductors are being operated within their specified range of conditions, and "infant" or early failures have already been weeded out. A plot of semiconductor failures versus time takes on the shape of a "bathtub" curve, as shown in Figure 3, beginning at some relatively high failure rate but falling steeply and quickly to a very low failure rate that is maintained for most of the semiconductor's life, finally rising up sharply at the end. Most manufacturers will burn-in a product a few days before shipping to catch the bulk of these infant failures.

The two major normal operation failure modes are contamination and mechanical flaws. Contamination-caused failures will occur as some function of on time, the reason being that such failures are usually the result of a chemical reaction between a foreign substance and the semiconductor. As we all remember from Chemistry 101, chemical reactions are accelerated by elevated temperatures. The reaction may not stop at room temperature but will slow down.

For those interested in a little trivia, the computer failure that scuttled the Space Shuttle launching a few months back was traced to a speck of human perspiration that fell inside one of the integrated circuits during manufacture. Months later (years?) the chip failed. Perhaps that explains why they use three identical computers on the Shuttle; two out of three are bound to work for the whole mission.

Mechanical flaws, a second common failure mode, are insensitive to on-time, and will be aggravated by on/off thermal cycles. Something like a weak wire bond on an IC pad will be stressed only so many times before breaking. There is an exception to mechanical flaws, indpendance of run time, and that is the case of a metalization path (sort of like a p.c. board run on the IC) which, because of the flaw, is too small for the current flow it sees. Eventually, metal will actually migrate down the trace, causing an open-circuit failure.

So far it's a toss-up. But there are other reasons why semiconductors fail. For some cosmic reason, coffee and soda seem almost magically attracted to the faders, while a static spark will run halfway across the console to find a weak IC. Likewise, power line spikes can make it into a console to degrade or blow-up chips.

There are also design-specific factors that can tilt the balance one way or the other. If a design has marginal thermal management (it gets too hot inside), and you turn off the air-conditioning at night, you could be asking for trouble leaving it running. Likewise, just turning your equipment off during a thunderstorm is no guarantee of safety; surges have been known to jump power switches!

Another consideration is the effect on passive components. While I am not aware of any that degrade with use, electrolytic capacitors will lose electrolyte faster at elevated temperatures. This phenomenon is likely to be more noticed in power amplifiers and power supply units where the designer doesn't have the luxury of using a capacitor several times larger than needed (because of size/cost), and there will be a build-up of heat during use.

There is no definitive answer that can be universally applied. Ask the manufacturer; ask other people with similar equipment; and finally use your common sense. If the studio is only down two hours a day, shut it down (and look for some marketing help). Either way, it doesn't hurt to have voltage transient suppression equipment in place.

PS. My personal opinion favors shutting the equipment down whenever possible, but, since I can't justify it with hard evidence, do whatever you prefer. There are good arguments for either choice.

PPS. The light bulb in my desk lamp just burned out when I turned it on. Hmm . . . ■■■

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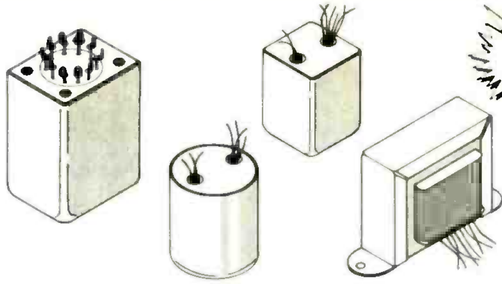

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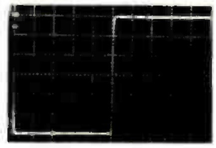

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MICROPHONE INPUT																
† JE-16-A JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	−0.08/−0.05	200	−8	<1	1.7	−30	1	A = 1 B = 2	64.21 68.86	42.89 45.99	29.60 31.74
† JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	−0.09/−0.21	85	−19	<2	2.3	−30	1	A = 1 B = 2	64.21 68.86	42.89 45.99	29.60 31.74
JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	−6	0.170/0.010	−0.50/+0.10	115	−5	<7	1.5	−30	1	3	42.03	28.07	21.92
LINE INPUT																
JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	−0.03/−0.30	52	−28	<3		−30	1	1	103.47	69.13	47.69
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	−0.03/−0.25	85	−23	<1		−30	1	3	40.05	26.76	20.90
† JE-6110K-B JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	−0.02/−0.09	125	−12	<1		−30	1	B = 1 BB = 2	62.86 71.52	42.01 47.79	30.83 32.97
* JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	−0.11/−0.08	160	−9	<2		−30	1	3	41.56	27.76	19.16
JE-11SSP-8M	Line in/repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	−0.03/−0.00	120	−9	<3.5		−30	1	4	151.90	101.47	70.01
JE-11SSP-6M	Line in/repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	−0.25/−0.00	160	−5	<3		−30	1	5	79.22	52.91	36.51
SPECIAL TYPES																
† JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	−0.16/−0.13	100	−12	<1		−30	2	3	34.60	23.13	18.06
† JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	−0.14/−0.16	100	−12	<1		−30	3	3	60.09	40.15	31.35
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	−0.10/−1.00	40	−18	<1		−30	4	1	96.90	64.73	44.66
JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	−0.20/−0.20	80	−18	<1		−30	2	6	43.57	29.11	22.73

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

PACKAGE DIMENSIONS:

	W	L	H
1 =	1$\frac{5}{16}$" Diam.		× 1$\frac{5}{16}$"
2 =	1$\frac{3}{16}$"	× 1$\frac{3}{16}$"	× 1$\frac{5}{8}$"
3 =	1$\frac{1}{8}$" Diam.		× 1$\frac{1}{16}$"
4 =	1$\frac{1}{2}$"	× 1$\frac{3}{4}$"	2$\frac{1}{2}$" w/solder terminals
5 =	1$\frac{5}{8}$" Diam.		× 1$\frac{3}{4}$"
6 =	1$\frac{1}{8}$" Diam.		× $\frac{15}{16}$"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot[8] (%)	Package[9]	PRICES 1-19	100-249	1000
* JE-123-BMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+28	2	−1.1	20 Ω	0.002/0.002	−0.02/−0.02	>450 160	−2.1 −4.1	<1	7	87.41	44.17	30.47
* JE-123-DMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+21	2	−1.0	19 Ω	0.004/0.002	−0.02/−0.00	>450 230	−1.2 −2.5	<1	8	50.71	33.88	23.38
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	−1.1	20 Ω	0.041/0.003	−0.02/−0.01	>450 170	−1.9 −4.0	<1	7	61.30	35.79	24.70
* JE-123-DLCF	Quadfilar	600-600 150-600	1:1 1:2	+27	2	−1.0	19 Ω	0.065/0.003	−0.02/−0.01	>450 245	−1.2 −2.5	<1	8	39.61	26.45	19.42
JE-123-SLCF	Quadfilar	600-600 150-600	1:1 1:2	+23.5	2	−1.1	20 Ω	0.088/0.003	−0.03/−0.01	>450 245	−1.2 −2.8	<1	9	33.48	22.35	15.43
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	−1.6	29 Ω	0.114/0.003	−0.03/−0.01	>450 205	−1.2 −3.2	<1	10	25.48	17.01	12.49
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	−1.3	8 Ω	0.125/0.003	−0.04/+0.06	190	−4.6	<6	8	42.14	28.15	19.42
JE-11S-LCF	Bifilar w/split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	−1.7	63 Ω	0.058/0.002	−0.02/+0.01 −0.02/−0.05	>10MHz 155	+1.1 −4.1	<1	8	42.14	28.15	19.42

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

† IMPROVED PERFORMANCE * NEW MODELS

PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7 =	1$\frac{1}{2}$"	× 2$\frac{5}{16}$"	× 1$\frac{15}{16}$"	2$\frac{13}{16}$"
8 =	1$\frac{5}{16}$"	× 1$\frac{15}{16}$"	× 1$\frac{5}{8}$"	2$\frac{3}{8}$"
9 =	1$\frac{1}{8}$"	× 1$\frac{11}{16}$"	× 1$\frac{3}{8}$"	2"
10 =	1$\frac{1}{16}$"	× 1$\frac{7}{16}$"	× 1$\frac{3}{16}$"	1$\frac{3}{4}$"

Prices shown are effective 6/1/84 and are subject to change without notice. Packing, shipping, and applicable sales taxes additional.

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

jensen transformers
INCORPORATED

For additional information circle #15

High Definition Audio

For the complete picture

The new 300 Series Audio Production Console has been specifically designed to complement the latest audio and v deo technology. It's the only console in its class, offering mono or stereo inputs each available with or without equalization, output submastering, audio-follow-viceo capability, a comprehensive user-programmable logic system, and a wide range of accessories for custom tailoring to your specific requirements. Available now. Call us collect for further information.

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3750 Old Getwell Rd.
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For additional information circle #16

January 1985 Number 4

ECHO TIMES

TIMEFLEX SURFACES AT A E S NEW YORK

Timeflex is a modified and specialised version of the DMX 15-80S dedicated to time compression or expansion. Whenever audio, film or video is speeded up or slowed down the dual de-glitched intelligent pitch changers within Timeflex can restore the correct audio pitch. The system is housed in a 2 Unit high rack mount case and is capable of either stereo or dual channel operation. Timeflex also incorporates the possibility of programmable delay offsets should sound/vision synchronisation be required.

Following the preview at AES New York and subsequent demonstration of the system at certain facilities immediately after the exhibition, A.M.S. received orders for the first 11 units for delivery in December '84.

REVERB PROGRAME BAR CODE UPDATE

The first issue of barcodes on laminated card for updating existing RMX 16 digital reverberators via their remote terminals were made available in mid October. This software update includes both new programmes and issues of programmmes previously released, but not currently available on REV 3.0 version software.

The initial list of programmes available includes: DELAY 8, DELAY 16, ROOM B1, FREEZE, ROOM A0 IMAGE P1, REVERSE 2, NONLIN 1, PLATE B1, HALL A1. Any three of these programmes may be stored at the same time in the soft programme locations, programmes 10, 11 and 12 in the RMX 16 mainframe. This first issue will be supplied free of charge to all RMX 16 owners when updating their remote terminals to accept bar code readers.

THE HOLLIES DROP IN AT A.M.S.

Tony Hicks and Bobby Elliot of the Hollies called in at A.M.S. to collect an RMX 16 and a DMX 15-80S which had been purchased by the band for both studio and live work. The Hollies had become familiar with A.M.S. equipment whilst working in studios both in England and America.

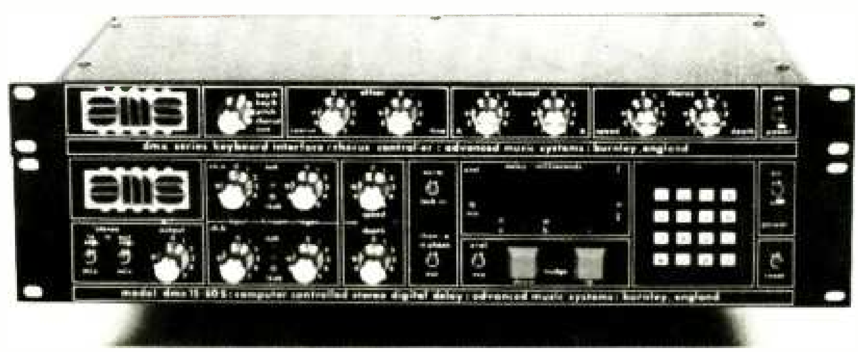
KEYBOARD INTERFACES NOW BEING SHIPPED

The first batch of 50 keyboard interfaces were shipped during October following the demonstration of the first prototype unit at the APRS exhibition in London during mid June.

The keyboard interface is a 1 Unit high addition for any DMX 15-80S DDL Pitch Changer allowing control of various functions by means of any keyboard capable of providing a 1 Volt per octave control voltage and Gate. The unit has currently four major functions:

1. Control of either the A or B channel pitch changers over their 2 octave range by means of any compatible keyboard.
2. Control of Loop Edited Samples over the same range.
3. Rotating pot control of both A and B channel pitch changers.
4. A new programme called Chorus which when selected automatically randomly varies both pitch changers about the unity setting, the user having control of both speed and depth varation.

The general feeling amongst the first owners of this low priced add-on is that it provides a wealth of additional facilities, particularly in the realms of manipulation of stored sampled sound, for the already versatile DMX 15-80S.



PEOPLE IN THE KNOW

"On Leave It I used a Linn and played that through an A.M.S. and delayed it for a full bar".

Alan White, drummer with Yes talking about tracks on the Yes 90125 album with Andy Hughes of International Musician.

☆☆☆

"Being a re-mix room and not a control room, the outboard selection is very special indeed. Offerings include the A.M.S. RMX 16 and 80S units which are almost becoming as essential as the new Dragonslayer Video-disc games!"

Chris Everard, Editor of sound Engineer magazine.

☆☆☆

"The thing that has been most exciting to me over the last few years has has been the effect of a good reverberation units. I like the sound of the A.M.S. better than anything else. I got terrifically excited when I first tried that out because it's beautiful – one of the deepest and best reverbs I've ever heard. I always look for a unit that gives me that perspective back behind the speakers - the Lexicon does a bit, but the A.M.S. has got real depth to it."

John Foxx in an interview to HSR magazine.

☆☆☆

"The A.M.S. is God. That's all you need to know. Totally the single most revolutionary thing I ever acquired was the A.M.S. digital delay. I've never had a session on my own stuff or anyone else's where my finger's off that button for more than half an hour."

William Orbit of Torch Song in an interview with Sean Rothman.

☆☆☆

"One of the things we need most where digital technology has really helped us, is to be able to put a voice in a room; considerable numbers of clients ask for the sound effect of an empty room. You tweak the old A.M.S., define a room of a certain size and suddenly - yes! - you've got a room. At first we were seriously concerned whether the cost of the unit would be recouped. Well now we've got four A.M.S. systems.

From the manager's point of view, the way to get the best employees is not only to pay good salaries; it's got a lot to do with the equipment the engineers are going to work with. I think it's absolutely certain that if an engineer was offered another job with a higher wage which meant switching from the latest good equipment to old rubbish, he wouldn't move. If they are working eight or nine hours a day in a studio the equipment is very important to keep the motivation high and maintain excitement and interest. If we had a staff meeting and said do we want five pounds a week more each or shall we buy a couple of A.M.S. harmonizers, another reverb unit or whatever I honestly think there would be no contest. Staff prefer new equipment and are willing to make five pounds extra a week on overtime when we get more bookings. Modern, high technology equipment enables you to attract and keep the best engineers.

Robbie Weston, M.D., Silk Sound – one of the U.K.'s four major companies in the field of audio for T.V. or Radio talking to Alvin Gold, International Broadcasting.

USA DEALERS IN THE KNOW

"A.M.S. is Hot: I've got a studio with three rooms who bought an RMX 16, they then bought another and they are used all the time. I even have a studio who've got all their existing outboard gear up for sale to finance the purchase of A.M.S. systems."

Nigel Branwell. Audio and Design/Calrec... A.M.S. dealer Washington State.

☆☆☆

"A.M.S. products are proven products universally well received in the marketplace. Each customer's evaluation has turned into a purchase."

Courtney Spencer. Martin Audio Video Corporation... A.M.S. dealer in New York City.

☆☆☆

"We feel the A.M.S. product range offers our clients the sound quality and ▷

THE FIXX

Pictured above are the Fixx whilst spending some time on the A.M.S. stand at the APRS exhibition in London. Being produced by Rupert Hine means they are no strangers to A.M.S. equipment and their time was valuably spent examining the keyboard interface for the DMX 15-80S. ■

KEN TOWNSEND TESTS THE DRIVING SEAT.

Pictured above is Mr. Ken Townsend, studio manager of EMI Abbey Road studios in London during a recent visit to the A.M.S. factories. Not unused to the pressures of management Mr. Townsend is pictured here behind the desk of sales and marketing director of A.M.S. Mr. Stuart Nevison. It should be pointed out that Groucho Marx look-alike Mr. Nevison had opted to remain incognito during Ken Townsend's brief spell of duty at the helm! ■

performance they are looking for in digital audio processing. What more can we say? – They must be great as they are selling like hot cakes on a cold winter morning!"

John Alderson. Studio Supply Company... A.M.S. dealers in Nashville.

☆☆☆

"Everywhere I take A.M.S. equipment producers and engineers get real excited when they get a chance to play with it... it really is fun to show!"

Ron Timmons. A.I.C. Company... A.M.S. dealer in Northern California.

☆☆☆

"A.M.S units are the hottest thing in town! The reasons people are buying them is the quality of sound, their versatility - there is a lot more to them than delay, pitch change and reverb - and of course their reliability factor. What's most exciting is we are really beginning to hear their effect on American productions as well as the European ones that they have dominated for so long."

Harry Harris. Harris Sound. ... A.M.S. distributor and dealer for the Los Angeles area.

HUMBERTO GATICA

Humberto Gatica arrived in the U.S.A from Chile in 1968. Thirteen years ago he accidently walked into a recording studio and knew that he wanted to be involved and by his own admission he has been very lucky and very successful. He has been involved in many projects including part of Michael Jackson's 'Thriller', and albums by Kenny Loggins, Fee Waybill, Dan Hardman, Kenny Rodgers and many more. Humberto enjoys his work and was able to confirm the rumour that he was recently spotted running between three control rooms at Sunset Sound studios in Hollywood - the simultaneous projects he was involved in at that time were Kenny Rogers, Kenny Loggins and Chicago!!

A.M.S.: What was it that alerted you to A.M.S.?

H.G.: I listen to a lot of other people's records and I really am a big fan of 'The English Sound'. I've listened to a lot of English records and there was definitely a sound that I considered unique. For instance, there is an English band called The Fixx and they have a source of delay they use that I really love - and I know you'll tell me it's A.M.S.

A.M.S.: The Fixx are produced by one of England's most fanatical A.M.S. users - Rupert Hine.

H.G.: Exactly! The same sort of sound that he used on the recent Tina Turner album - and the best way I can describe it is 'unique'. So for three years I've been using A.M.S. units and there isn't a session when I don't use them in one way or another.
The most important thing I can say about A.M.S. is that the products are very musical. I can be recording a synthesizer and the musician will say - "Hey, what are you doing to the sound? It's fantastic! - I can't believe it!" - and my easiest explanation is - "Oh... just using a little A.M.S." There is always something in the recording process that if put through A.M.S. units makes it sound better. I recorded a Christmas album for Kenny Rogers and Dolly Parton where many of the A.M.S effects used were very subtle - but, if you took the A.M.S. effects out even when used subtly you really could tell the difference.

A.M.S.: How do you like to use the RMX 16?

H.G.: I am fanatical over a drum sound. I will use 'Necam' to remove every drum back beat from a snare so all I am left with is the impact of that snare - now feed that to the RMX 16 reverb and the effect is awesome and well worth the time spent. I really like the brightness of the RMX 16 - take the Ambience programme, it's clean with lots of top end which means when you bring the music up in a mix it's still there - that's just not the case with all digital reverbs. I also like very much the Nonlin and Reverse programmes. I recently completed a new Chicago album and there were several cuts where I used the Reverse programme on the brass sound - and they loved it. It really added a new dimension to the sound. For the past 17 years their horn section has been a major part of the record and what do you do if somebody asks you to give a new sound? Somehow a short decay setting on the Reverse programmme does!

A.M.S.: What about strings?

H.G.: I find it impossible to explain what I want from strings, but by using the RMX 16 I can place the strings exactly where I feel they sound best. Taking an RMX 16 everywhere with me I can go into any studio and hardly worry about the room. I can make a big room sound small and a small room sound big! A.M.S. units are everything I need in a mix because they make it so easy to create depth and place for not only strings but everything. I believe it is important to make a record feel like everyone is there and playing at the same time and A.M.S. units are my biggest help in creating this feel whether it be delay, echo, pitch change or reverberation.

A.M.S.: And how about the DMX 15-80S pitch changers?

H.G.: A.M.S. have literally changed the whole business with their units. Everywhere I go I make fans for A.M.S. - Julio Inglesias is a very sensitive man and to him the vocals are the most important part of his recording. The last time we worked together it was in America and he just wasn't happy with the result because with American musicians he had been forced to sing right on top of the beat. He was very precise when recording but afterwards we could play around with the DMX 15-80S and program it so that each phrase he sang could be 'layed back' just sufficent for him to feel comfortable again with the end result. And he was right, it made a lot of difference. Julio fell in love with the A.M.S. and was looking for a unit to take everywhere with him!

A.M.S.: Do you use the Loop Editing System?

H.G.: There is so much you can do with A.M.S. units but there again there are many things I feel I haven't fully taken advantage of yet. I started work on Quincy Jones's new album and when we sat down to talk about it Quincy was very excited about getting very heavily involved with this new sound - and of course he's talking about A.M.S. sampling! I am a really big fan of A.M.S. and the best way I can put it is - A.M.S. makes recording fun.

ELECTRIC LIGHT ORCHESTRA

In the very early days of A.M.S. it was thought that the DM 2-20 Flanger, because of its stereo outputs and dual channel analog delay function, could successfully replace rotating speaker cabinets for use with electronic keyboards. E.L.O. were the first band approached and asked to consider the DM 2-20 as a valuable piece of equipment for their use. Both Jeff Lynne and Richard Tandy of E.L.O. are now both A.M.S. converts and carry their own units wherever they go.

A.M.S.: After your introduction to A.M.S. when did you next come across any of our units?

Richard Tandy: Following our first meeting, which I do remember, I next encountered A.M.S. systems at Ridge Farm. What immediately excited me was the quality of the DMX 15-80S - particularly the bandwidth of the system.

Jeff Lynne: The quality just made it so difficult to use anything else. They are all fabulous, we really do just get so blasé and it's not unusual to hear a shout of " just put another 2 hours of delay on this for us!" - confident that that will not cause any problems.

A.M.S.: Between you what do you really make best use of when you are working with A.M.S. units?

R.T.: For me, I do work a lot with drum machines and when writing, the accuracy of the programmability of the DMX 15-80S has made it so easy for me to get the sound I want.

J.L.: I am a big fan of the RMX 16 reverb. I really do like the Ambience programme if only because it is so obviously designed to have very little colour. I love either very short decay settings or very long ones.

A.M.S.: We have talked to people who write tunes around pieces of equipment. Do you work like this or not?

J.L.: No, not really. I do write a song by trying to get a good tune first - that's the most important bit. What's really nice then is that I always find that A.M.S. comes into how the arrangement works and they really are such a pleasure to work with at that very important level.

A.M.S.: So do you feel A.M.S. plays an important role for you?

J.L: There is no question that A.M.S. really did change our lives! Sampling using the Loop Edit System is amazing. I've actually done a Christmas Record for my friends - it features my father and we really should send you a copy - it's brilliant. There really is no point explaining it to you now because you would never get away with publishing it in any respectable magazine!!

VISUAL MUSIC SCENE

Three-Dimensional Sound for Video: Discovery Music Network to Utilize Holophonic Audio for 24-Hour Music Video Channel

by Adrian Zarin

The stage is set for Phase Two of the Video Music revolution to commence in the first months of 1985. In Phase One, as you may recall, the form, content, and consumption patterns of the Visual Music medium were established, to a large extent, by a single force: MTV. The cable network's hegemony as the only 24-hour source of Music Video made it a monolithic power that overshadowed all other forms of Visual Music programming. Its influence was (and is) so pervasive that it even changed the nature of its non-visual sister media, records and radio.

The coming year, however, will see the advent of several new 24-hour Music Video cable channels. Among these are entries from the powerful Turner Broadcasting System, and the Discovery Music Network. The latter operation is being launched by Les Taylor and Karen Tyler, who first blazed cable trails with their Financial News Network. While the original video boom was aimed at the youthful 12- to 25-year-old demographic, these new cable networks offer a CHR/Top-40 format targeted to the more mature 24- to 45-year-old group. The cost of home video hardware and software being what it is, it has become clear that this older group constitutes more of a "video generation" than their young counterparts, who often find it hard enough to come up with the price of a record album. (In recognition of this fact, MTV has also launched a second 24-hour music channel, VH-1, also aimed at the older audience.)

With all of these contenders in the field, competition for viewers' attention naturally will be keen. For the Discovery Music Network, *audio* will play a large role in attracting viewers. The company recently announced that it will be broadcasting 100% of its program material in Holophonic™ sound. Zuccarelli Communications, the originators and sole proprietors of Holophonics, has set up a base within Discovery's West Los Angeles headquarters, which contains production studios to be used for the music channel and on-site satellite uplink facilities.

According to Brian Adams, head of Zuccarelli's licensing group, the Holophonics company "entered into an arrangement with Discovery through their parent company, Amnet, whereby they have provided us with certain facilities here, including demonstration rooms and office space, and we're working with them on the broadcast and transmission of holophonics to as broad an audience as possible."

Holophonics, as many *R-e/p* readers will know, is a sort of audio equivalent to holographics. By recreating the psycho-acoustic processes of actual human hearing, according to the Zucca-

— STOP PRESS —

At press time, it was reported that the Turner Broadcasting System's Cable Music Channel has been purchased by MTV Networks, Inc. for a claimed $1 million. The agreement involves the transfer to MTV of CMC's subscription list, and also calls for the purchase by MTV of $500,000 in advertising time on other TBS networks — *Editor*.

← AMS FOUR-PAGE INSERT PRECEDES ON PAGES 27 to 30

Discovery
MUSIC NETWORK

relli people, holophonics creates a "three-dimensional" sonic image. Not only can the listener precisely locate the spatial source of sounds anywhere along a two-dimensional plane (front-to-back, and left-to-right), he or she can also hear sounds as though they were emanating from above or beneath the listener's location.

Holophonic Principles

Acoustic theorist Hugo Zuccarelli is the inventor of Holophonics, a process that reportedly grew out of his dissatisfaction with previous models of human hearing, which stated that the brain uses information from both ears to locate sounds in space. Noting that he was able to accurately locate the source of sounds with one ear blocked, Zuccarelli devised an alternative theory.

"Zuccarelli feels that the ear is an active participant in locating sound," explains Ken Caillat, the Fleetwood Mac co-producer who has signed on as head of Zuccarelli Communications' Licensee Support Group. "He believes that the ear actually has its own reference tone which is always present as a sort of background noise — a sound that the ear recognizes as constant. It's like the undisturbed surface of water in a swimming pool. When a sound enters the ear, it disturbs the reference tone and actually sets up an interference pattern exactly like the interference pattern in a hologram.

"Based on his theories, Zuccarelli developed a model of the human hearing system, which is what we are now marketing. To the best of my knowledge, it seems to exactly duplicate what a person actually hears."

Holophonic sound can be recorded and played back on conventional audio equipment. The only difference is the conventional microphones are replaced by Zuccarelli's holophonic recording devices, a black box he has whimsically dubbed "Ringo." The exact contents of Zuccarelli's black box is one of the darkest secrets in the audio world; it's a subject that the Holophonics people are curiously reticent to discuss.

"I don't exactly know everything that's inside it," confesses Caillat, "but basically it attempts to duplicate how we hear. In a recording situation, we would go in, listen to the recording environment and material to be recorded, and then place our device or devices in the best spot or spots depending on the application we're going for."

Vee Jay segments for the Discovery Music Channel, interviews with artists and other links between clips will be videotaped and holophonically recorded at the Discovery studios. The audio recording will be a simple matter of placing one or more Zuccarelli pickups on the set with the Vee Jays and other personalities. Discovery plans to script and stage these segments in a manner that will take full advantage of holophonics surround-sound properties.

"It will be cute to have somebody like Dolly Parton or Stevie Nicks saying, 'Hi, here I am,' and then coming over and whispering in the listener's ear, walking around in back of him . . . that sort of thing," says Caillat. "We really will be able to establish a greater feeling of personal contact with the individual viewer. With holophonics, he will hear exactly what he would hear if he were on the set with the guest star."

Audio Processing

Imparting the Holophonic effect to audio tracks of video clips will involve their being re-recorded with one or more of Zuccarelli's pickups. This can be done using the stereo audiotracks on the videotape furnished by the record company. The Zuccarelli people, however, say that they can achieve more dramatic effects by using the song's multi-track masters. In either case, the audio would need to be played back through a loudspeaker array and re-recorded holophonically.

. . . continued overleaf —

Discovery
MUSIC NETWORK

"People will no doubt ask, 'What can you do with just two channels?'" Caillat offers. "We have a matrixing system that will enable us to grab the high frequencies and 'wiggle' them a little bit, rotate them and otherwise move them around. But that's admittedly somewhat limited.

"What we're hoping is that, as the word about holophonics spreads, there will be people who will say, 'Isn't there more you can do? What if we come into the studio with you and remix the multitrack tape putting the snare drum over here, the guitar up there . . .?'

"With the multitrack master to work with, we could pick out a particular instrument, send it over your head and down your back on a particular passage leading to a chorus, for example. Holophonics can add a whole new dimension to mixing."

"We want artists to get involved," adds Brian Adams. "We want their creative input into how holophonics can be applied to music. If they work on a clip with us and like the result, maybe we can work on their next record with them and come up with an entire holophonic album."

The re-recording of audio tracks for clips to be aired on the Discovery Music Channel will all be done digitally, although the holophonic effect can also be obtained with analog tape. "We feel that holophonics is going to have a big impact on digital audio by subtantiating why digital exists," Caillat comments. "It becomes *more* than just a matter of having a cleaner recording. With holophonics, you can really appreciate digital's low noise floor, because you are dealing with a perfectly realistic image of acoustic reality."

Apart from clips and studio segments, Discovery also plans to eventually offer the full gamut of specials and concert telecasts — all, of course, accompanied by holophonic sound. The cable outlet even plans to offer its facilities for the production of holophonic commercials. While no definite production plans have been made at this time, the Zuccarelli people point out that there are several differing options for recording programs and commercials holophonically.

A concert, for example, could be recorded "live to holophonic" with one or two Zuccarelli pickup devices. If preferred, however, individual tracks could be recorded (or re-recorded) holophonically. As another option, a conventional multitrack recording could be "remixed to holophonic" by playing back a mix through speakers, and re-recording it with a Holophonic pickup.

Domestic Playback

Zuccarelli Communications further claims that, if you holophonically record music that is being monitored through a set of high-end studio monitors and then play back the holophonic recording on a modest set of five-inch speakers, the small speakers will precisely reproduce the sound heard over the studio monitors. How, one might ask, is this possible?

"Brain information," responds Caillat. "Because holophonics is an accurate model of human hearing, the brain says 'I am hearing the studio monitors.' *That* is the information the pickup device has 'heard,' and that is what is passed along."

Which brings us to the subject of playing back holophonic sound in the home. It can be done on conventional home audio equipment with completely satisfactory results, say the Zuccarelli people. "You don't have to buy any kind of decoder," Caillat explains. "There are no rear-channel speakers or surround system to be added to your home equipment either."

The biggest problem in holophonic playback, apparently, is phase cancellation. Rooms containing a lot of hard, reflective surfaces do not make good holophonic playback areas because sound waves bounce around and mutually cancel one another, diminishing the holophonic effect of precisely locating sounds in space.

Similarly, multiple-component speaker systems can present problems as well, because the individual speaker components are likely to be out of phase.

"Everybody knows that a three-way speaker has a ridiculous amount of phase shift," says Rich Feldman, also of Zuccarelli Communications. "As you move through a room, your position relative to the tweeters, mid-range and woofer speakers is constantly changing and you're therefore getting phase shift. Manufacturers have known that for a long time. Only now are they starting to go to point-source, phase-coherent speakers. Sony, Technics, Cetec Gauss and Tannoy have all come out with models."

The only equipment acquisition the Zuccarelli people would recommend for holophonic playback at home is a set of phase coherent, point-source speakers, which are less expensive, in many cases, than their three- or four-way counterparts. But even this investment isn't an absolutely necessity, since some holophonic effect will be apparent on any playback system. Also, if the listener owns a pair of stereo headphones, he already has an ideal Holophonic playback system.

Ken Caillat recommends placing the speakers at an angle between 40 and 90 degrees from the central listening point. "It seems," he observes, "that if you are located dead center between your two speakers, you will get a totally even balance between front and back information. Unfortunately, that position makes for a narrower window than if you position the speakers at a 45-degree angle. You will loose a little bit of rear information that way, because you will actually be in the rear a little more. At the same time, though, eight people can sit there and hear the holophonic effect."

As part of the programming material offered between videoclips, Discovery plans to run segments that combine entertainment and holophonic education. Guest celebrities will demonstrate to viewers the best way to place their speakers, and basic principles like phase coherence will be explained. "We feel that one of the main benefits for us of going up on satellite with the Discovery Music Channel is the education about holophonics that we can give to 20 million people all at once," says Discovery's Rich Feldman. "We hope we can achieve something like what has happened with computer technology. Everybody understands terms like 64K or 128K today, because they've been saturated with information about computers. There's no reason why the average person can't develop the same type of sophistication when it comes to audio."

Even the home viewer that has no more audio equipment than the proverbial three-inch speaker mounted in an inexpensive television set will be able to hear the holophonic effect, according to the Zuccarelli people. "It should sound pretty good on a home TV set," says Caillat. "The kind of speaker you would encounter there is a traditional point-source, phase-coherent speaker. You will definitely get some up and down, some right and left and absolutely some depth information. Holophonics seems to work very well in mono. Our research has shown that even a person totally deaf in one ear can hear sounds moving in space, and can hear the holophonic effect perfectly."

Just as holophonic sound doesn't require two speakers, the Zuccarelli people go on to explain, it also doesn't require that the listener remain stationary at a precise point between two speakers in order to perceive the holophonic effect. As in a hologram, the spatial illusion holds up from any vantage point.

"Sure, there *is* a sweet spot where you'll have to be if you want to get the exact perspective that was recorded," Feldman clarifies. "But that's not the *only* perspective to be heard, nor necessarily even the most desirable one. If you are listening to music, you might not even *want* to hear the same perspective that was recorded. You might want to walk around the room, move among the instruments and pick out a spot that you like better!"

Another peculiarity of holophonic sound that makes it particularly attractive for Music Video telecasts is that it can't easily be pirated. A tape copy of a

Holophonic transmission made on consumer equipment would not contain the full sense of spatial dimension as the original. Tests that Zuccarelli Communications made with consumer recording equipment have shown that the holophonic effect degenerates on such equipment just like frequency response, and other parameters that diminish when taping conventional audio sources.

Holophonic sound seems perfectly able to survive transmission intact, however, a fact that has been confirmed by other tests Zuccarelli Communications made in conjunction with the cable outlet. The Discovery Music Channel will transmit its holophonic audio signal via satellite, and will reach viewers either directly by cable or by means of FM simulcast, depending on local facilities.

Digital Transmission

All programming will originate at the Discovery complex in West Los Angeles. Using their own satellite dish and Wegener Communications uplink equipment, Discovery will distribute two separate stereo signals (one analog and one digital) as video subcarriers. (Negotiations regarding which particular satellite will be used were still underway at the time of writing.)

"Wegener Communications has a very high-quality digitizing system, which we will be using to uplink one audio signal," explains David P. Chandler, Discovery's vice-president/ engineering. "We don't expect a lot of people will be using that signal initially, but it will be there. What we are also going to do is uplink a conventional analog signal, which can be used by those stations which already have that kind of equipment. The truth of the matter is that the analog signal will provide quality that is as good as they can get through most of the local transmission systems."

Along with this, Discovery will transmit a third audio signal to be used in FM simulcasts; this signal will be transmitted from Discovery to IDB, a nearby uplink facility, which will beam the signal to RCA SatCom 1R for general distribution.

"We've tested the holophonic effect with the RCA and Wegener systems, and have broadcasted it via standard FM," says Chandler, "and it seems to muscle up to just about anything."

"The only thing that I'm not 100% sure of are the 'ifs' of the particular station — the transmitter and matters like frequency roll-off and limiting," adds Caillat. "How things like that will affect the holophonic sound will vary from station to station. But even under the worst of circumstances, holophonics will provide better quality sound than conventional audio. The listener will get better dynamics, more sense of spatial movement, and more volume for less amplification."

One is always a little skeptical of technological developments that claim the power to revolutionize the audio world. After all, there has certainly been no shortage of false alarms and failed messiahs in the past. But after listening to Zuccarelli Communications' demonstration tape, this writer was convinced of holophonics' ability to create dramatic acoustic effects, and particularly to create the illusion to sounds moving through space.

Precisely how the "3-D" realism of Holophonic sound will complement the "2-D" fantasy of Music Video remains to be seen. While it may seem something of a novelty or luxury item at present, it's importance could increase significantly with the advent of home television — a development which some observers feel is not *that* far away. It is clear though, that if the spatial illusions of holophonics can be applied to music recordings and successfully transmitted to home audiences, they would be likely to give the Discovery Music Channel a pronounced edge over their competitors. Discovery and Zuccarelli Communications, moreover, are convinced that this "edge" will be particularly meaningful to their audio-conscious target demographic, the Woodstock/*Rolling Stone* generation weaned on high-tech hi-fi ads.

It *would* be odd if a cable television channel became the catalyst for the next revolution in audio, which the holophonics people seem profoundly convinced they are offering. But looking back on MTV's far-reaching impact on the non-visual media, it's apparent that a precedent already exists. ■■■

TRAVELS WITH THE EDITOR

Digital Developments at the New York AES Convention

by Mel Lambert

With the professional audio industry now utilizing digital recording and production technology at an ever increasing pace, The recent AES Convention in New York proved to be a valuable venue at which to catch up with binary developments. In the top-end markets, Digital Entertainment Corporation reports that it has now placed a total of eight Mitsubishi X-800 digital 32-tracks in studios around the country — including recent installations at Clinton Recording, New York, Audio Effects, L.A., and Digital Associates, Nashville — and that over 50 X-80 digital two-tracks are now gracing the control rooms in just about every major recording market throughout the U.S. Sony also appears to be making good progress with sales of its DASH-format PCM-3324 24-track, with a total of 24 transports delivered, including recent purchases by Glen Glenn, Hollywood, CBS, New York, Dallas Sound Labs, Texas, Hit Factory, and Blank Tape, New York, Bruce Botnick's Digital Magnetics, Hollywood, and a third multitrack for Village Recorder/MRI, Los Angeles.

Possibly the hottest news on the digital recording front was the U.S. unveiling of Studer's D820 DASH two-track. Utilizing the same basic transport design as that featured in the new A820 analog two-track, the D820 is available in two- or three-head configurations (the latter enabling digital read-after-write for off-tape monitoring), and features selectable 44.1 and 48 kHz sampling frequencies, plus AES/EBU inputs and outputs for direct connection to digital consoles and processors. All transport functions are under full microprocessor control for gentle tape handling and fast location to any point on the tape. As with the A820, audio and transport parameters can be reprogrammed under software control, enabling special functions to be set up via a secondary keyboard and assigned to soft keys. A combined shuttle bar and cue wheel provides precise control of tape position during manual editing and cueing. Deliveries of D820s are expected to begin in the second quarter of next year; pro-user prices will be in the region of $20,000.

Sony also is expecting to begin shipping of its PCM-3102 DASH two-track by mid-year. I understand that the reason for the delay in market availability of both the Studer and Sony transports stems from recent enhancements to the DASH 7½ ips, quarter-inch two-track format, following technical meetings between Sony/MCI, Studer, and Matsushita. DASH specifications that have now been ratified include symmetrical layout of the 12 data tracks (eight digital data, two cue, and one each for timecode and reference data), and optional PWM encoding of the cue-track information to improve replay quality during editing. (While Sony appears to be opting for a "conventional" analog cue track, Studer DASH two-tracks will offer selectable record/replay of PWM or analog cue data, to ensure compatibilty between tapes recorded on either transport.)

On the digital console front, Neve was showing the DSP Disk Mastering Console destined for delivery in early 1985 to Tape One Studios, London. A full description of the DSP's features, along with a report on my recent visit to DSP-equipped CTS Music Center and Tape One will be included in next month's "On the Studio Trail" section of *Studio Update*.

Also on show was Sony's new eight-channel Digital Mixing System, which accepts a combination of analog, direct digital, or 1610-format inputs, and provides three types of output: stereo main, stereo subs, and eight directs. In essence, the system consists of three subunits, the K-1106 eight in/out A/D conversion unit, K-1107 eight in/out D/A conversion unit, and the K-1105 Signal Processor and Control Unit. As a result, for CD mastering and digital mixing of 1610-format material, all that is required is the K-1105 processor and console "control surface," which features full equalization, level and pan adjust, plus effects and monitor assigns. Designed primarily for CD mastering duties and high-quality digital recording to stereo, shipment of the Digital Mixing System is planned to begin early next year.

If the upper-end market is currently being dominated by Mitsubishi, Studer, Sony, JVC, 3M, and other manufacturers of 16-bit recorders and processors, the potentially more cost-conscious section of our industry is rapidly coming to terms with 14-bit (and 16-bit) EIAJ digital processors. Even though Sony and similar firms may have postured such devices for consumer use, there is no denying that the PCM-F1/701, Nakamichi DMP-100 and related processors are finding applications on more and more sessions.

And, if only to show that Sony isn't the only manufacturer of EIAJ-format processors, JVC recently unveiled the VP101, which offers BNC video inputs and outputs, and built-in sync input port; projected price is $895. Also available from JVC is the FX900 digital conversion unit, which digitally transcodes 14- or 16-bit EIAJ material to VP-900 16-bit format, and includes CRC, error concealment and mute condition indicators.

Ease of operation, reduced tape and equipment costs aside, working with EIAJ-format digital material is not without its own inherent problems. In particular, because of the way in which the digital bitstream is encoded onto successive video frames, the editing of EIAJ-format material is not a trivial exercise. While it's relatively easy to transfer material in the analog domain from an EIAJ processor to a JVC VP-900 or Sony PCM-1610 16-bit device for more precise digital editing, there exists a potential degradation in signal-to-noise, frequency response, and distortion of the transferred audio. (Not to mention the higher editing costs with VP-900/1610 systems.)

Among the various companies addressing the EIAJ-format editing dilemma, Audio+Design/Calrec, Editel/New York, Harmonia Mundi Acustica, and KEMA Marketing were showing some particularly interesting hardware at the New York AES Show.

A+D/C offers three stages of "professional modifications" for the Sony PCM-701ES processor: Section One comprises electronically balanced, line-level XLR inputs and outputs, plus CTC (Coincident Time Correction) circuitry to remove the 11.34-microsecond delay introduced between the left and right channels by the processor's single A/D converter; Section Two includes digital inputs and outputs, video sync capability, and switchable copy inhibit and pre-emphasis, plus the optional Ad-Mix Digital Fader that enables level adjustments during transfers to be made in the digital domain, as well as the ability to add a second digital signal to the one being replayed off-tape, and then record the composite signal onto a second VCR (in essence, the digital equivalent of "sound-on-sound"); and Section Three, which comprises an interface for direct digital-to-digital transfer of the 701's bitstream to 1610 format, and vice-versa, plus remote control of CTC, emphasis, copy inhibit, and NTSC/PAL format. Pro-user prices are

. . . continued on page 41 —

A+DC's modified Sony PCM-701ES

For additional information circle #21

Why do the world's leading studios turn to Solid State Logic?

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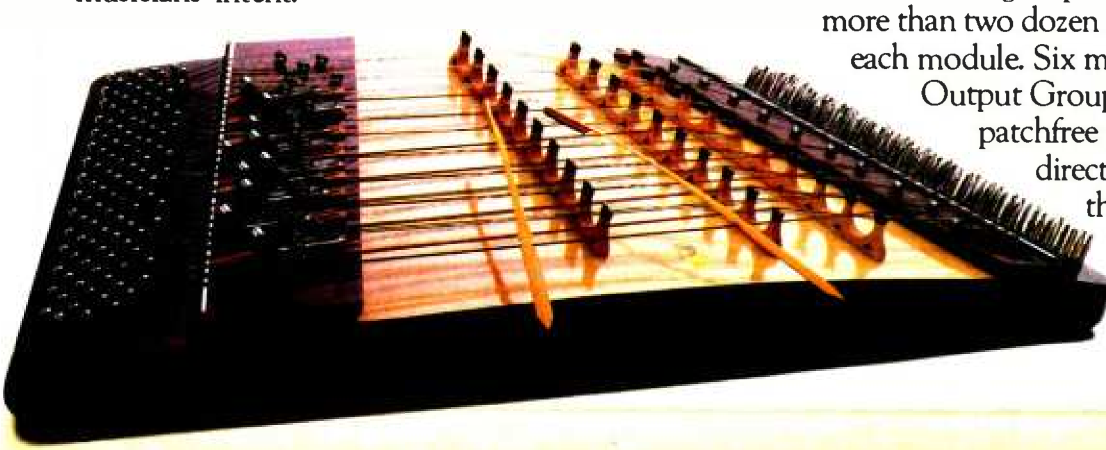

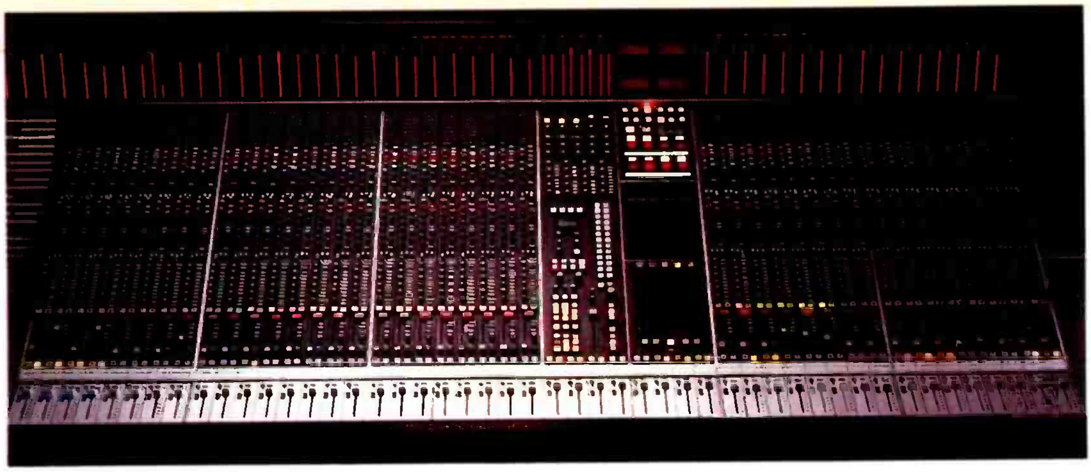

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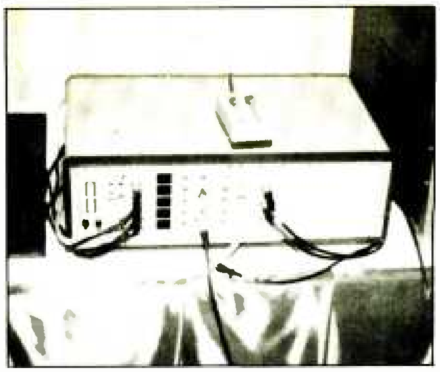
as follows: Section One $480, Section Two $525, Section Three $875, and Ad-Mix around $1,000.

Also designed for direct digital transfer from F1/701 to 1610 (optional 1600) and AES/EBU digital format, the new BW102 Digital Audio Interface from Harmonia Mundi Acustica GmbH, of West Germany, utilizes internal 24-bit processing, and enables the following signal modifications to be made entirely in the *digital* domain: high-pass filtering (selectable 1, 15, 60 and 120 Hz LF rolloff); DC-offset cancellation; left/right channel reversal; independent level change and phase inversion control for each channel; plus correcting L/R delay time offset, pre- and de-emphasis. The unit is entirely modular, and can be set up with any combination of input/output and processing modules; two or more frames may be connected together for special applications. U.S. distribution of the BW102 will be handled on the East Coast by Audiotechniques, in the Midwest by Allied Broadcast, and on the West Coast by Audio Intervisual Design. Pro-user price is expected to be in the neighborhood of $8,000.


Harmonia Mundi
BW102 Digital Interface

The two EIAJ-format editing systems on show at the AES Convention ranged in complexity from a simple assembly editor developed at Editel/New York by David Smith, to HHB's Computer Logging Unit and Editor (CLUE) for automated control of editing VCRs, plus off-line compilation of edit decision lists.

A functional prototype of Dave Smith's D/S-1000E Editing Co-processor, being demonstrated on Audiotechniques' stand, consisted of a Nakamichi DMP-110 with the additional editing circuitry built into the unit's battery compartment. (The basic design principles of the ECOP, including internal conversion of the EIAJ video interleave block format was described on page 81 of the October issue of *R-e/p*.)

To perform insert and assembly edits of F1/701 material, the record VCR utilized with the D/S-1000E must be capable of video insert editing — for example, a Sony SLO-383 Betamax or BVU-800 U-Matic — and the video editing system based on control-track operation (such as the Sony RM-440 PAC controller), or a timecode synchronizer (Audio Kinetics Q.Lock, BTX Shadow/Softouch, Adams-Smith, etc.), or a CMX Video Editor. Edits can be made to the nearest video frame (33.3 ms), which, as Smith readily concedes, may not be accurate enough for tight music editing, but is perfectly satisfactory for film and video sessions. At present, the prototype ECOP is capable of "performing 85% valuable edits," he offers, "although we are working on ways of improving that figure, and expect to have an improved version available by late '84."

Somewhat more upmarket in terms of control ability, the CLUE System, developed in England by HHB Sales, and available in the U.S. through KEMA marketing, a division of AMEK Consoles, Inc., is a fully integrated F1/701 editor that enables butt copy-editing to be performed in either the analog or digital domain, to an accuracy of one video frame. And where increased accuracy is required, CLUE is said to significantly reduce the amount of time required on a more accurate (and expensive) bit editor, such as the Sony DAE-1100, by providing off-line EDL preparation.

Clue provides comprehensive logging and autolocation facilities, as well as more accurate tape-location counters than those found on most domestic-style VCRs; the unit will also function with standard U-Matic recorders. The only modification needed to connect an F1 or 701 to the system is a single additional wire to provide a control-track output from the processor for timing purposes.

Designed by HHB's David Wilkins, who spent over two years developing the system, CLUE enables up to 200 edit locations per tape to be logged and stored to the unit's 5¼-inch floppy disk drives; up to a total of 99 sets of tape-location data can be recorded on a single floppy. During the recording or mastering session, the system allows the user to log details of each take, including track title, start location, duration, etc., as well as the position of 10 locations during the take (to label, for example, false starts, performance errors, solos, and so on). During playback an additional 10 locations can be annotated. Autolocation is achieved by inputting an absolute counter number, or by reference to a take number or a logged location, thus doing away with the need to input timecode numbers. The program software will also calculate lengths of sessions, takes, or musical passages.

HHB's CLUE F1/701
Editing System

The complete CLUE system comprises a 4U 19-inch rack unit that houses the controlling computer, disk drives, switching circuitry, and interface connections, plus a remote ASCII keyboard. CLUE also provides full access to audio and video outputs of the record and playback VCRs, as well as analog inputs and outputs of the F1 or 701. A front-panel analog fader allows level correction to be made during mastering or editing sessions. Future developments for the system include additional interfaces for audio and video recorders, as well as the ability to read and write timecode. U.S. pro-user price is expected to be in the region of $8,000: two systems reportedly have been sold to the BBC, plus single systems for Harmann Digital, West Germany, and Stewart Copeland of the Police, for use in his personal-use 24-track studio.

And if your digital manipulation requirements are more complex, the recently developed DSP-2000 Series of digital recorders and mixers from CompuSonics is being promoted as an extremely flexible and interactive editing system. Stressing once again that conventional magnetic tape represents an ideal long-term storage medium for digital audio data, CompuSonics president David Schwartz offered that "the real advantage [of the DSP-2000] is its random-access and editing capability. It is *not* economical to use hard-disk to store an hour of 32-track digital material. Our system is designed to be used to load the material onto hard disk for editing and manipulation, and then to be stored on data cartridge or recording tape."

The latest generation DSP-2000 enables 20 minutes of 16-bit mono audio to be stored onto a single 140-Mbyte hard disk; up to eight disk drives can be accessed from the central controlling computer. It must be realized, however, that CompuSonics is still developing the system, and has high hopes of increasing the storage capacity — through a combination of data compression and variable sampling frequency — to a maximum of 30 minutes of stereo (or 60 minutes of mono) in the not-too-distant future.

A basic DSP-2000-2 system, which occupies two, seven-by-19-inch rack-mount units, consists of a stereo input/output unit; a 140-Mbyte fixed hard-disk drive; a 3.3-Mbyte "super floppy" 5¼-inch disk drive for temporary or archival storage and program loading;

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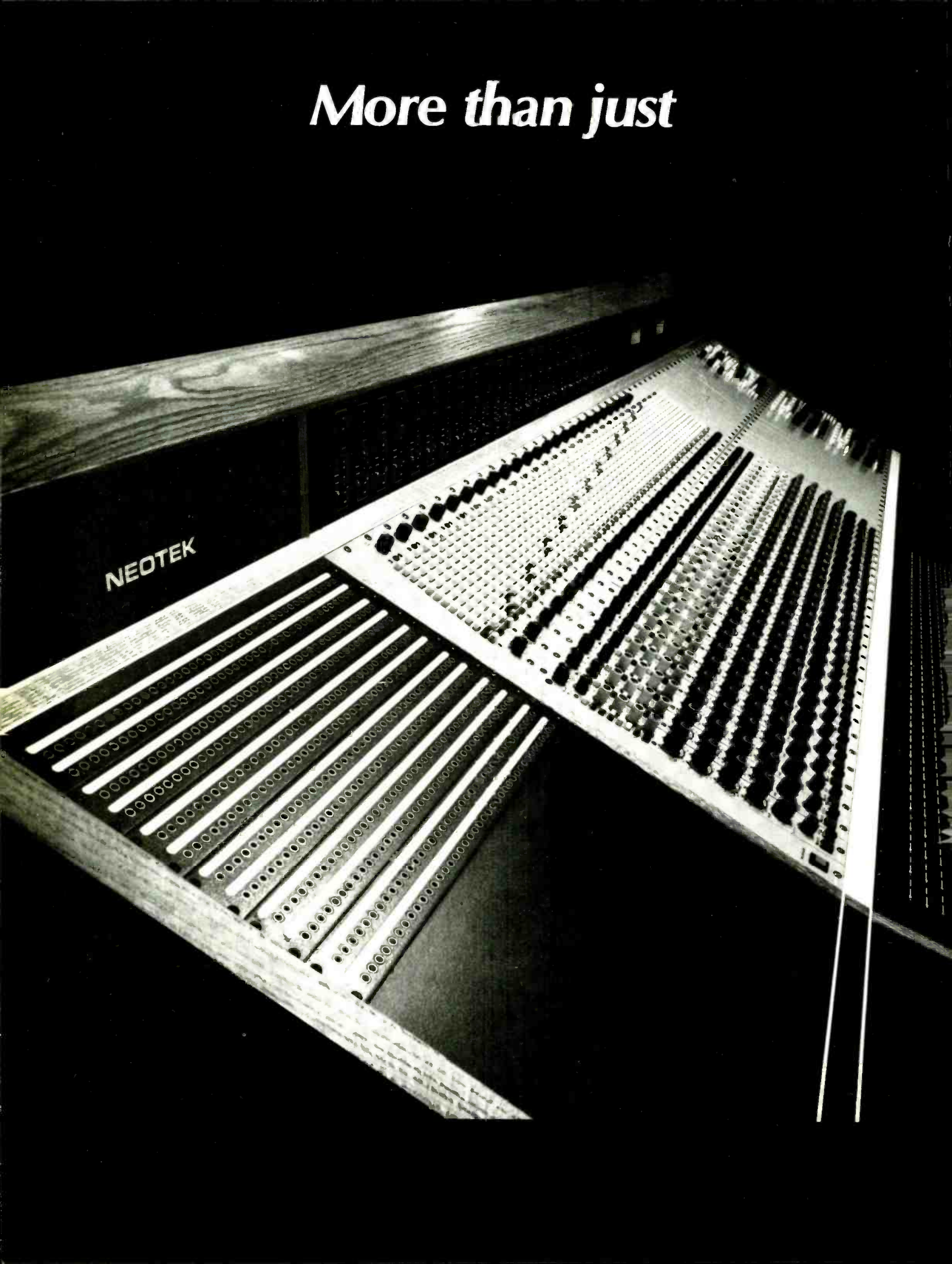
More than just
NEOTEK

a pretty face

NEOTEK consoles have the look and feel of serious professional equipment. Heavy structural extrusions, eighth-inch thick front panels, all metal patch bays, and solid hardwood cabinetry are complimented by a control layout that is easy to reach and easy to use. You may not care what's inside that makes them sound so clean, quiet, and correct . . . or perhaps you do.

Minimum path design NEOTEKs employ more complex circuit designs than other consoles, yet they end up with fewer components in the main signal paths. Their state variable equalizers are highly sophisticated, yet critical parameters are less sensitive to component variations than in simpler designs, they are just as quiet, and they are far more stable; the result is better sound at only slightly higher cost. Compared to a Series III, one comparable console when recording and mixing passes a track of audio through 53 more op amps, 49 more unbypassed electrolytic capacitors (none in the NEOTEK), and wastes over 106dB more excess gain. Of course the NEOTEK sounds better!

High speed circuits The rule for circuit speed is 1 volt/microsecond for each peak signal volt. NEOTEK consoles are the only ones which come close to this figure; others trade adequate speed for lower parts cost. The real trick is to achieve high speed without slew limiting, but since NEOTEKs use circuits with power bandwidth in excess of small signal bandwidth, they can never be forced to actually slew. Full output bandwidth from mic preamp input through equalizer and fader to bus out is over 40 kHz, and high frequency squarewave response shows purely exponential signal rise and fall without a trace of slewing, ringing, or other instability. The result is absense of TIM or SID, greater stability, and the clear, sweet high end which distinquishes NEOTEK consoles.

Solid state switching The Series III uses FET switches for master status control, but fear not. They are a unique design using discrete devices driven from a separate high voltage supply. At the last AES show, an internationally famous audio critic and recordist guessed that a B&K mic demo was made direct to digital two-track; it was actually made on a Series III . . . now that's transparency. The ramped FET mix mutes silently lift channels completely off the stereo buses. They are far more quiet than VCAs, relays, or mechanical switches, and leave the unweighted output noise below-96dB.

Logic controlled mutes NEOTEK consoles provide full professional features without requiring an automation system or VCAs. Series IIIs provide two logic groups for the channel mutes. There is an in-place solo mode in addition to stereo and PFL solos, and it can effect either or both groups. Another logic system sets up a limitless number of grouped mute/unmute events to be enacted by a single switch.

Subgrouping without VCAs VCA grouping is an unnecessary added cost for the Series III. Switches on each input allow panning to eight stereo subgroups while adding eight additional auxiliary buses to the standard six. The subgroup masters, which also serve as stereo line inputs, have auxiliary sends, logic mutes, and individual echo returns. Unlike VCA subgroups and automation, this system allows use of a stereo compressor on the entire group. This subgrouping system coexists on automated NEOTEK consoles.

Have it your way NEOTEK manufacturers a full range of consoles designed for specific applications. There are console series for four-and eight-channel recording, multitrack recording, broadcast production, theater effects and sound reinforcement, film and television post production, and sophisticated sound reinforcement. Each is built to individual order in the United States. Engineers at the factory are available to tailor each console to the most demanding applications.

If you are about to choose a console, choose NEOTEK.

Let others compromise.

NEOTEK CORPORATION

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— continued from page 41 . . .

TRAVELS WITH THE EDITOR

a controller/power supply; and CRT and keyboard. Projected prices range from $34,800 for the basic 2000-2 configuration, to $55,800 for the 2000-X4, which includes three additional hard-disk drives to provide additional data storage.

Currently, the 2000 Series features one Mbyte of internal memory, which provides approximately 10 seconds of storage capacity for locating exact edit points. The system is configured to operate with a 50 kHz sampling frequency, which in future versions will be automatically adjusted by the processor to suit the frequency range of the material being sampled. In this way, the sampling frequency can be dropped to 25 or even 12.5 kHz in the presence of material that consists of predominately low-frequency information. CompuSonics says that on average — and this is a parameter that obviously depends on the type of material being sampled — the unit can provide up to three times its present capacity through the use of an "adaptive" sample frequency. (Hence the projected total storage time, when the variable-Fc circuitry is implemented, of 20 × 3 minutes of mono audio.)

Also on display at AES was a prototype CompuSonics four-input stereo mixer that features trackball controls for each level and EQ function. Still under development, the mixer provides a full-color display on a VDU of EQ, foldback, pan and level settings, and in future versions will have built-in automation capability. (Hence the use of trackballs as control surfaces, since they do not need to be reset to a particular position, plus the fact that they operate in two directions — to provide, for example, simultaneous control of EQ cut/boost and center frequency, or level and pan.) CompuSonics recently appointed the following pro-audio dealers for its range of digital consoles and editor/recorders: Audiotechniques for the East Coast; Allied Broadcast for the Midwest; and Audio Intervisual Design for the West Coast. ■■■

MIDI UPDATE

A Report of the MIDI-Equipped Outboard Equipment and Synthesizers On Show at the New York AES Convention

by Bobby Nathan, Unique Recording Studios, New York

Now that the dust has settled from the New York AES Convention, at least one thing is apparent: MIDI in the studio is here to stay. In fact, seeing MIDI in/out jacks on outboard gear makes you wonder, "What next?" Included below are details of some of the more interesting MIDI-equipped outboard processors and synths on show at the AES.

The new **Publison Infernal Machine 90** is a dual-channel special-effects unit, with each channel capable of functioning as either a mono-in/stereo-out digital reverb, five-second digital delay, a glitch-free pitch shifter, or a digital sampler with up to five seconds of sampled audio. The software developed so far will allow an engineer to use the two MIDI In jacks — one for each channel — for two powerful features. The first enables the unit to be interfaced with any MIDI-equipped keyboard to control the pitch-shift interval. For example, you could play the lead vocal track through one channel of the Publison and bounce it to another track of the multitrack with the appropriate minor or major third harmony (or perfect fourth/dominant fifth, etc). The second feature allows the device to sample a vocal "ah," for example, and play melodies from any MIDI keyboard. Since the unit comprises two, totally independent units in one chassis, both these effects (or any of the other combinations) could be produced simultaneously.

AMS has now added an interface box for its **DMX-1580s** dual-channel delay/pitch-shifter/sampler. Connected to the DMX, the new Keyboard Interface/Chorus controller provides the user with the ability to select either channel A or B, and control the interval of that channel's pitch shifter. The controller will also let you play the "sampled" sound at different intervals in "Loc" mode, or enable the pitch rate and chorus depth to be controlled randomly for both channels A and B from an external control voltage. Although no MIDI ports are yet provided on the interface — only CV and Gate In — possibly in the future a MIDI In jack will become a reality.

Another interesting item of MIDI-equipped outboard gear is the new **Yamaha D1500**, a digital delay that has preset capabilities for storing your favorite delay parameters — including delay time, regeneration, LFO and mix settings — in the unit's internal memory. When interfacing the D1500 to a Yamaha DX-7, or any other MIDI compatible keyboard that sends preset patch information, preset patches on the synthesizer and delay settings can be changed under MIDI control at the same time. This feature enables particular delay settings to be associated with synthesizers sound patches. It would seem likely that some day soon, we will have the ability to automate delay settings from the multitrack, allowing for a different delay for the chorus than the verse, and so on.

MIDI Out from your studio's acoustic grand piano, Yamaha CP-70/80, Fender Rhodes or Wurlizter? **Forte Music** has figured it out and, yes, it works! Installation is said to be relatively quick, and with an indiscernible change to the acoustic qualities of the above mentioned instruments. After your instrument has been "Midified," you can control any number of MIDI keyboards at once. As well as note-on/note-off information, the velocity of the modified instrument's keyboard is transmitted through the MIDI port; even the keyboard's sustain pedal is MIDI capable, and will sustain all synthesizers equipped to read this date. (For example, Yamaha's DX-7, Roland's Super Jupiter, E-mu Systems Emulator II, etc.) The real splendor of this breakthrough can not be realized until you've played it or heard it!

Garfield Electronics (of Doctor Click fame) has added the MIDI Adaptor sync box to its line of interface units that allows MIDI Clock devices to be synchronized to existing machines via clock pulses (96, 48, 24 beats-per-quarter-note, Roland sync, etc.). Garfield has also added an FSK adapter box to add "sync-to-tape" capabilities to many MIDI Clock and DIN sync devices (by KORG and Roland). There are also over a half dozen other new, lower priced interface boxes, designed to solve specific problems. (Of course, where would we be without the Doctor?)

The new **Korg KMT60** is a one-in/six-out, buffered MIDI Thru Box that reduces any delay caused by patching into one synthesizer and out via its MIDI Thru jack into the next, etc.

J.L. Cooper Electronics, manufacturers of over a dozen different devices for slowing MIDI interface problems, has come up with yet another novel idea, **The MIDI Patchbay**. This 16 channel-in/eight channel-out patchbay includes LEDs to show routing status of the 16 inputs, and a microprocessor capable of remembering 16 patch presets. The box should prove particularly useful for interfacing several MIDI keyboards, sequencers and/or "On-screen" synthesizer editors, and is configured to allow quick changes of what device is controlling what (master/slave status).

Speaking of "On-screen" editors, **Yamaha's DX Pro Software** for the Apple II, II+, or IIe is now available. If you've had trouble editing sound patches on a DX-7, your worries are over. Not only is each operator's envelope displayed separately, they can be simultaneously superimposed over each other. Another great feature is the ability to

. . . *continued on page 171* —

Grammy-winning producers edit with JVC Digital Audio Mastering System.

Marc Aubort and Joanna Nickrenz, of Elite Recordings, Inc., New York, widely known and highly respected producers, were awarded a 1983 Grammy for "Best Producer of the Year, Classical" for six records: three on the Nonesuch label, and three on the Moss Music, Inc., label. All six were recorded & edited with the JVC DAS-900 Digital Audio Mastering System.

"In classical editing, time is of the essence because budget is always such a critical factor," says Marc. "Classical music is often heavily edited, a painstaking process that eats up time and money. JVC enables us to combine manual cueing and memory edit with a speed we couldn't believe."

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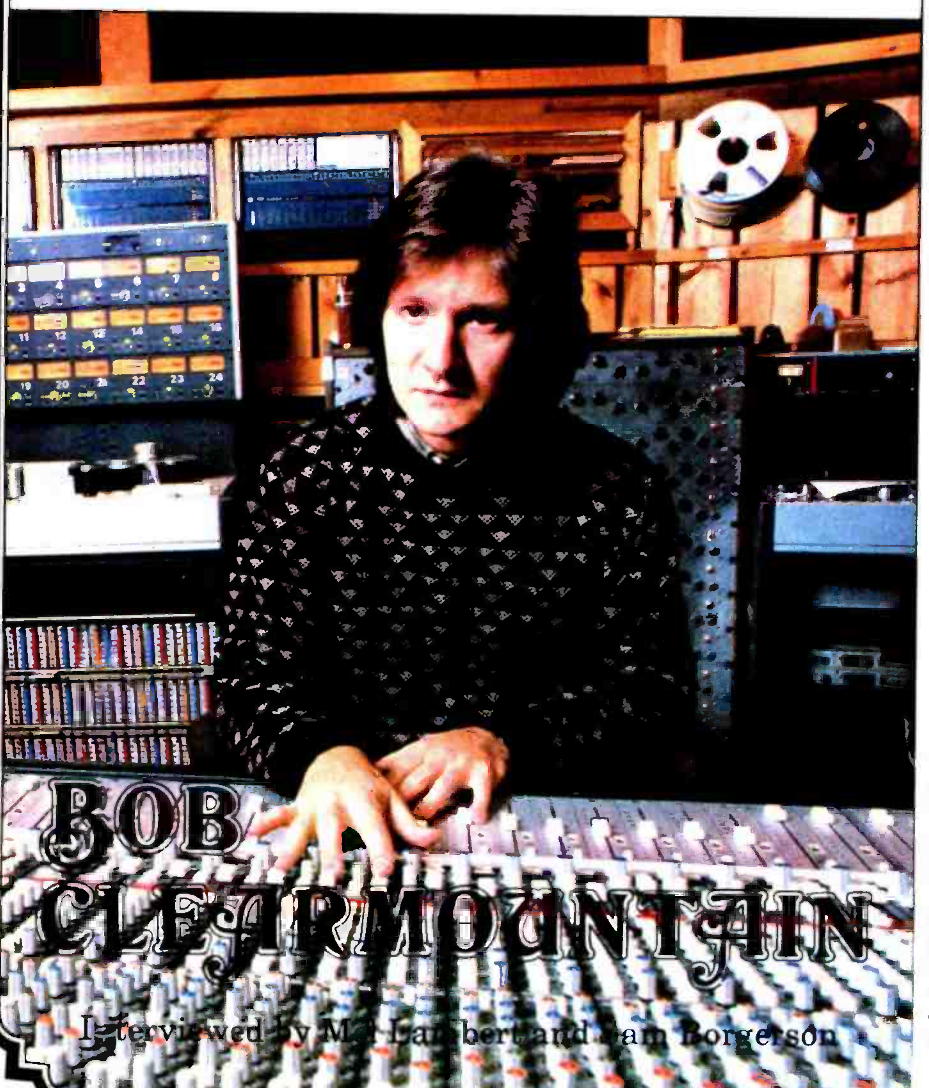
PRODUCTION VIEWPOINT

Drawing on a rare combination of keen intuition, refined musical sensibility and patient perfectionism Bob Clearmountain has quickly advanced to one of this industry's leading mixing engineers. In 1984 alone his mixing and co-production credits read like a Who's-Who of rock royalty: Hall & Oates, Bryan Adams, Bruce Springsteen, Huey Lewis and the News, Mick Jagger, Ian Hunter, and Little Steven (Steve Van Zandt). In between his album mixing projects, he has somehow found time to handle a premix of music for film (The Rolling Stones' *Let's Spend the Night Together*); HBO Video Specials; and live *King Biscuit Flower Hour* concert radio broadcasts for DIR. By sheer weight of albums sold (not to mention widespread critical acclaim), Clearmountain is shaping the sound of rock'n'roll in the Eighties. Not a bad showing considering that, in the early Seventies, he was a struggling bass player for several now-defunct Connecticut bar bands.



BOB CLEARMOUNTAIN

Interviewed by Mel Lambert and Sam Borgerson

Photography by Sam Seligson

R-e/p (Mel Lambert): Did you have any technical training before you entered the recording industry?

Bob Clearmountain: No, except I had been playing with tape recorders ever since I can remember. My mother was an English teacher, and would record her students. I can remember back when I was about five years old, she would bring the recorder home. "Hey, lemme see that! What is this thing here?" I've been intrigued by recording all my life, so it seemed the obvious thing to explore the studio scene. I had dabbled in electronics, and I was definitely into music.

We were in the process of doing some demos at Media Sound when the band I was in finally broke up, so I started hanging out there. Michael Delugg, the engineer there, was real nice to me, and he gave me a good plug to the owners. I just kept coming in and annoying them until they hired me!

I worked as a go-fer for an hour and a half; I did two deliveries. After I came back they said: "Where have you been? You're not a go-fer, you're an assistant engineer. You're supposed to be down in studio A." I'm 19 years old, I walk in, and it's a Duke Ellington session! This engineer I don't know is telling me to go out there and move microphones around.

R-e/p (Sam Borgerson): Had you done any engineering or mixing before that time?

Bob Clearmountain: No, except with my own bands, and with friends. I had a little Radio Shack four-channel mixer; one of those little boxes with four knobs on it. But I got pretty sophisticated with it. I actually rigged up a little talkback into the other room. But it was pretty basic; we didn't even know how to edit.

So I was amazed when my band was being recorded at a real studio and, when we made a mistake, the engineer said, "Okay, just take it from the bridge." I had *no* idea what he was going to do. Then I saw him cut the tape and splice it together. I thought it was a miracle, of course. I immediately went home and started recording everything off the radio and chopping it up, making the announcers say stupid things! I got into it *instantly*.

R-e/p (SB): How long did you work as an assistant at Media Sound, New York?

BC: It's hard to say. I assisted for about three years, but there were a lot of sessions where I served as first engineer. They were pretty generous to me. Actually, about two months after I started at Media I was assisting on a session for Kool and the Gang and, for some reason, the engineer wasn't particularly into it. He was a jingle guy, and just got booked on it. So he let me go for it. I mixed one song, and did a whole bunch of the overdubs. This was after two months, so I was really nervous.

. . . continued overleaf —

BOB CLEARMOUNTAIN

R-e/p (ML): Did somebody at Media Sound run you through the board's features, to get you started?

BC: Oh yeah. It's one of those situations where, if you don't ask questions, nobody's going to tell you anything. They tell you what they want you to do, but not what it all means. So I used to ask a *lot* of questions. Also, because Media was primarily a jingle house, the engineers weren't all that concerned about experimenting with new sounds. They got good sounds, but were more concerned with speed and efficiency. So, as an assistant, I would always be setting up different mikes to see how they would sound.

Eventually I would start suggesting things to the first engineers like, "Why don't you try a little 12k on the strings?" and some of the engineers who were nice guys would say, "Oh all right, I'll try it."

Of course now that I'm doing the records, I suppose I'd be upset if somebody switched mikes on me. But I do take suggestions; I like an assistant to ask me to try something. The assistant might have worked with somebody and picked up a good tip. I find it good to stay open to ideas.

When I go into a studio that I'm not familiar with, I'll usually ask the assistant what works in the room . . . where do you put the drums? I'll go with what the assistant is used to first, and if that doesn't sound the way I want it, try something else. I learned a lot by engineers letting me do things like that.

At Media Sound, about six months after my first session with them, I was working with Kool and the Gang again. The engineer, a different one this time, got sick and didn't show up. I'd set up the session before I knew he wasn't going to show, and — like it has happened many times before — I ended up doing the session. Two hit songs came out of it: "Hollywood Swinging" and "Funky Stuff." That was back around '73. Well, "Hollywood Swinging" got up to number six, and that was the first session I had actually recorded all on my own, so I was pretty pleased about that. Of course I didn't do anything that big for about two years afterwards.

R-e/p (SB): Did you do the mixing on the Kool tracks?

BC: No, I didn't. An excellent mixer named Jeffrey Lesser mixed it. It was actually good that I didn't because I didn't know anything about mixing at the time.

R-e/p (SB): When did you start becoming involved with mixing?

BC: Soon after that. I started to get pretty good at it the last couple years I was at Media Sound, which was '76 and '77. I left Media when Tony Bongiovi started Power Station — I was actually the first person he told he was going to do it, even before his partner, Bob Walters. Tony was one of the engineers at Media, and I had learned a lot from him. He also turned some R&B and disco clients on to me. Having had a couple of hits with Gloria Gaynor, and he produced that *Star Wars* record with Meco, Tony had some money with which to build a studio. He gave me a good opportunity to help design the place; it was mostly his design, but he let me put some ideas into it. One of the things I wanted was to have real *live* rooms.

R-e/p (ML): Back in the mid- to late-Seventies, this must have been several years before other studio owners woke up to flexibility and potential of live rooms.

BC: We were actually used to a live

sound, because Studio A at Media Sound was quite live. I was always into that room — they had a live room and a dead room; I *hated* the dead room. The Power Station room is live, but it is also quite versatile — it has sliding glass doors so you can divide it into three large, different-sounding areas. As a result, you can have live-sounding drums, piano and loud guitar with pretty good isolation.

R-e/p (ML): Let's talk about how, during the recording process, you work toward the final mix. I realize there are some obvious aspects to be taken into account during tracking, like setting good levels, trying to leave yourself space for bouncing, and keeping the good takes. But, beyond that, are you listening for certain sounds as you work toward the mix?

BC: Yes, I'm always working toward the mix. However, the thing that is in my head more than anything when I'm recording is the *performance.* I first concern myself with the basic sound, and get that out of the way. But lately I really go for the performance, because now the only sessions I record are those that I also co-produce. I know that if I get a good performance and the sounds are decent, there's going to be no problem. Obviously the arrangement has to be right but, hopefully, there's been enough pre-production so we don't have to worry much about that.

R-e/p (ML): If you're going to be overdubbing certain instruments, during the tracking dates do you have an idea where the "holes" will be later?

BC: Yes. When I'm recording I'm *always* trying to do a mix, which is why there are some studios I prefer to work in — and some I don't. For example, on a [split console] I don't know what to do because the monitor fader is somewhere down in the next county! If I don't have my hands on the mix faders I'm thrown off, and it's hard for me to get a feel of what is going on. It's also partly why I engineer the things I produce, although I'm thinking of getting out of engineering because [combining engineering and producing duties] is getting to be too difficult. But I *do* like having my hands on the faders, and having some echo effects going.

R-e/p (ML): Your various sound textures are coming together as you are recording the basics and overdubs?

BC: Yes, although it's not really a complete picture of the final mix, because usually it will end up different than you first visualize it.

R-e/p (ML): Do you try to leave a decent interval between finishing the tracks and mixing?

BC: I try to. If I can, I like to leave a week or two between finishing the album and mixing. Also, it's very difficult for me, if I have to record something in the day and then mix it that same night — or mix anything that same night.

R-e/p (ML): Is that because you are listening for different nuances during the mixdown stage?

BC: Yes. With mixing you want to get an *overall* picture; you don't want to tune in too much on one thing. When you're overdubbing, you're focused in on specific performances — with the mix

. . . continued overleaf —

"I just try to get the drums to sound as punchy as possible. The only basic rule is that there's a certain balance between the bass drum and the snare drum I always go for, and also a certain balance between the bass guitar and the bass drum. The blend of those sounds is very important."

you want to step back and listen to the *whole* record.

R-e/p (SB): You've been very successful working strictly as a mixing engineer, and handling the final mix on projects engineered by others. Do you approach those mixes differently than projects you might have engineered yourself?

BC: First of all it's a lot easier for me to mix tracks I didn't engineer, provided they are recorded well. Again, it's hard to get a good perspective on something you may have worked on for months. You've heard it *so* many different ways, and you know every little thing that has been played on every track. It's difficult to be objective about what is *important* to the song.

When somebody brings in a tape I've never heard before, the first thing I do is push up the faders and get a rough mix, then listen to the song a couple times. I'll solo things once in a while, just to get an idea of what the instruments are doing. But I'll listen to the vocal mostly, to get an overall attitude. Then I'll break it down and start building the mix. Sometimes I will listen to a rough mix they've done, but I find it's better not to do that — I prefer to get the multitrack and just start pushing up faders.

R-e/p (ML): You don't find that listening to a rough mix is helpful while setting up to mix a track?

BC: No, because that will have a certain set attitude. Sometimes an artist will want some sort of feel that they have in their rough mixes, but usually they'll want me to just go for it.

R-e/p (ML): Do you ever receive tapes for mixing, and push up the faders only to discover the tracks don't sound too good? Would you ever advise them to go back and try again?

BC: It does happen sometimes. But it's difficult to advise them to do more work because of the time and the money. Sometimes I'll just sit there and say to myself, "Boy, I wish this could have been better." But then it's my job to fix it; to make it sound as good as I can.

R-e/p (SB): Do you ever confer with the tracking engineer before a project if you know you'll be mixing it? For example, the Huey Lewis projects?

BC: No. Basically, his engineer [Jim Gaines] is very good, although he has a completely different style from me. He goes for more of a fat, dead, "Memphis R&B-kind" of sound. But for Huey that works, though it can't be *too* fat and *too* dead, because he's an exciting artist. It needs some liveliness in it, so I try to add that. But the tracks always sound great; he's an excellent engineer. I've talked to him a couple times, but I never say anything about how the tracks were recorded. I enjoy getting tracks recorded differently than I would have done them. It keeps me more open-minded about my own recording technique.

R-e/p (ML): And it must also expose you to a wide range of material, with different approaches?

BC: Right. That's one of the things I like about the position I'm in now. I don't like doing several long projects one right after the other. I like to break it up by just doing mixing. Especially for the next couple of months, because I did two very long projects: Hall and Oates [*Big Bam Boom*] and Bryan Adams [the new album *Reckless*]. I had a great time on both projects, but I'd like to balance that out. I'll be mixing a Bryan Ferry album in another month, and I'm sure Huey will have another one coming in pretty soon.

R-e/p (ML): You've worked with Bryan Ferry and Roxy Music before, haven't you?

BC: Yes, I've done two Roxy Music albums, *Avalon* and *Flesh and Blood*. When it comes to just pure mixing, his material is my favorite — they give me so much space to work with. The music sort of swallows me; I'll be mixing for a couple hours and hardly realize where I am, because I'm just swept away by lis-

Bob Clearmountain's track sheet and notes for slave and master multitracks to Hall and Oates' "Some Things Are Better Left Unsaid" sessions.

SLAVE

TITLE "Some things are better left unsaid" PROJECT Hall & Oates CLIENT Whole/Oats Prod

DATE 7-7-84
TAPE 3M-250 N/R: None
LEVEL +5/185 SPEED 30 ips

TRACK	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
PROGRAM	LV	Lead* Vocal	Lead* Vocal	V4	V5	SYNTH Swells	BV	BV	V's		GTR	R ARPEGGIOS	Lead Vocal	Bass	Claps	BD	DR	UMS	DX7	T-Bone GTR	DA GT	RYL RS	DR CLICK	smpte
COMMENTS	Possible Good Band AD LIBS	Verses & CHORUS use for "OUH" @ TAG	TAG	TAG 2	Verses V2 Tag 1 [illegible]	DX7 & JP8	Lo DN	HI V (use 9 & 10)	x4 Comp	x4 of 1 thru 3	@ TAG		Comp tracks 2-5 NCE	← Monitor Mix →										
MIC													FET-U47	10 PAD										
EQ	See	CN 13	for Mic EQ, ETC!										Gain: 40 Neve EQ -1 @ 15K +2 @ 330 HP @ 47											
OUTBOARD						Pcm42 315 ms FB @ 7							Pultec [illegible]											
MIX SUGGESTION									PHASING or FLANGING				252 ms DLY (1/4 note)											
N/R																								
STUDIO	A	A	A			B	B	B					A											
DATE	8/27	8/27	8/27			8/16	7/30	7/30					8/22											
ENGINEER																								

MASTER

TITLE "Some thing are better left unsaid" PROJECT Hall & Oates CLIENT Whole Oats Prod

DATE 6/29/84
TAPE Scotch 250 N/R: NO
LEVEL +5/185 SPEED. 30

TRACK	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
PROGRAM	BASS	END BASS	BD	[illegible]	Simmons [illegible]	CLAPS	BD	SN	KI L	T R	RO L	OM R	SLN [illegible]	Fairlight Bridge	Simmons Fairlight [illegible]	Chorus & BRSN	[illegible]	Guitar Fills	12 String Heavy Fairlight	Chords Strings	12 String Guitar [illegible]	[illegible]	CLICK	Smpte
COMMENTS	DX7	[illegible] FADE	LINN Turn off @ TAG	LINN	BKWDS CYMBL INTRO [illegible]	LINN	BKWDS SN Echo IN Verses	SN hit @ Verses MI	CKEY @ TAG					TBEE IV	B Words Cymb @ Intro [illegible]	Simmons Fairlight [illegible]	Intro Verses & Bridge	T-Bone Telecaster Roland Comp [illegible]	DA	RYL	Daryl [illegible] Track 5		DR CLICK TEMPO 119 BPM	
MIC																		57 [illegible]			[illegible]			
EQ																		No Board [illegible]			[illegible]			
OUTBOARD																		[illegible]			[illegible]			
MIX SUGGESTION																		[illegible]			Roland Jazz Chorus			
N/R																								
STUDIO		B																						
DATE	6/29	8/2	7/4 & 3		6/29										7/3	7/3		7/			6/29	6/29		
ENGINEER																					*SEE BACK	OF* SHEET		

The master multitrack was recorded at Electric Lady Studios, New York, with guitar, synth, and vocal overdubs to the slave multitrack at Power Station a week later.

Kenny Rogers' Lion Share Studio loves the new

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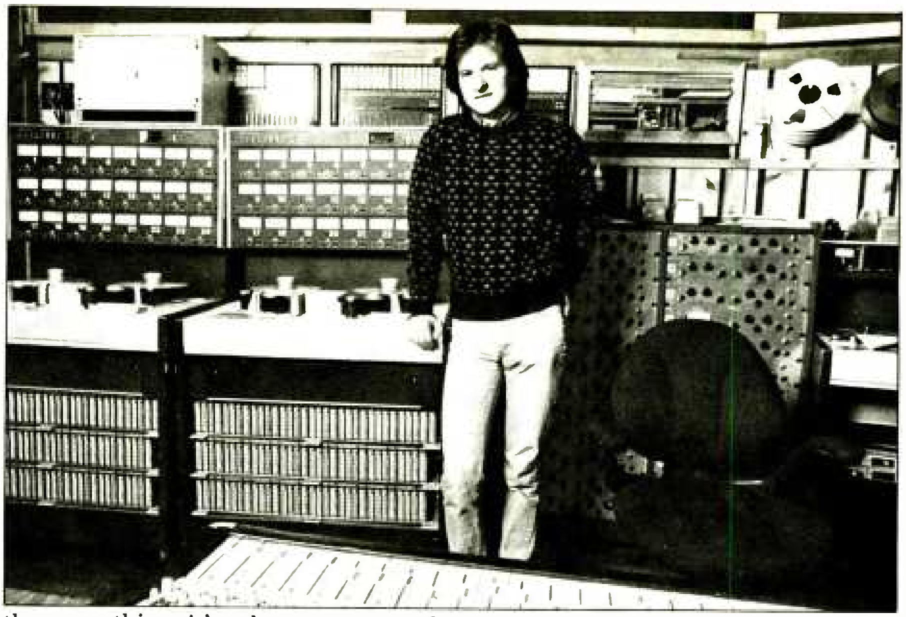
tening. And then somebody will tap me on the shoulder and I'll say, "Wow, who put in all of those patch cords?" It's brilliant music, and they're great to work with. Bryan's an extremely sensitive, understated person; a real gentleman.

R-e/p (SB): It might be interesting to contrast Bryan Ferry with another artist you've mixed recently, Steve Van Zandt. Ferry suggests an open and spacious sound, where Van Zandt is intense — a wall of sound. How do you come to terms with these different personalities, and decide on the right kind of sound?

BC: With Bryan Ferry it always seems obvious, and works out naturally. With Steve Van Zandt, it's a lot harder for me to tune into what he's doing, because it really is *very* dense, and there's not much room to work in. With him it's more a matter of squeezing it all into the record. I use all the little tricks of compression and limiting and EQ, but more than anything it's also a matter of arranging. You have to figure out what the important things are, and then underplay what isn't important. If you want to maintain a workable mix, it is much better to underplay the things that aren't important, rather than try to shove up the important things. It's like a photograph: Something has to be in the background to give the depth and perspective. But it's hard for me to analyze how I go about establishing this in the mix, because I do it instinctively.



PARTIAL DISCOGRAPHY

Album Productions

Hall and Oates/*Big Bam Boom*/1984
Bryan Adams/*You Want It, You Got It* /1981
Bryan Adams/*Cuts Like A Knife*/1983
Bryan Adams/*Reckless*/1984
Tuff Darts/*Tuff Darts*/1978
Narada Michael Walden/*Dance of Life*/1979
Garland Jeffreys/*Guts For Love*/1983

Album Mixing Credits

Roxy Music/*Flesh and Blood*/1980
The Rolling Stones/*Tattoo You*/1981
The Rolling Stones/*Still Life*/1982
Ian Hunter/*You're Never Alone With a Schizophrenic*/1979
Huey Lewis and the News/*Picture This*/1982
Huey Lewis and the News/*Sports*/1983
David Bowie/*Let's Dance*/1983
Bruce Springsteen/*Born in the U.S.A.*/1984
Little Steven/*Voice of America*/1984
The Divynls/*Desperate*/1983

Singles Mixing Credits

The Rolling Stone/"Miss You"/1979
Roxy Music/"Dance Away"/1979
Roxy Music/"Angel Eyes"/1979
Bruce Springsteen/"Hungry Heart"/1981
The Clash/"Rock the Casbah"/1982
King Crimson/"Sleepless"/1984
Ian Hunter/"I'm A Teacher"/1984

Other Mixing Credits

The Rolling Stones —
Soundtrack premix of the film *Let's Spend the Night Together*
David Bowie and Hall and Oates
Home Box Office Specials
"King Biscuit Flower Hour" DIR radio shows for Bryan Adams, David Bowie, and Duran Duran □□□

R-e/p (ML): Do you set up a mix visually, placing some sounds down in front, some in back, and then space them across the stereo spectrum?

BC: Yes, I do, although not to the point where I can draw you a picture. Sometimes I will set up the mix as a stage sound. Bryan Adams would be a good example: I'll have the drums in the middle; put the bass in the middle to get it on the disk; then the keyboards to one side; and maybe background vocals to one side as if the bass player was over there singing. Instead of putting the lead guitar in the center, I'll put it off to one side, because usually in the band he is off to the side.

Sometimes I will try some dynamic panning, if it doesn't come off sounding too gimmicky. I did one thing on the new Hall and Oates album where the lyric goes, "The dealer changes directions." There's a guitar lick after it that goes from one speaker to the other, the guitarist being the dealer. I occasionally pick up on a lyric cue, but you have to draw the line and not make it *too* obvious.

R-e/p (SB): While on the subject of dynamic mixing, what kind of control-room monitors do you normally use?

BC: I have these Yamaha NS-10Ms, which I guess are getting pretty popular. I've been using them for about four years now. An engineer at Power Station heard them at Motown in L.A. and liked them, so they got a pair out here. I put them up and thought, "These will never work, they sound *too* good." But, just for the hell of it, I did some rough mixes, took them home, and they sounded right, except they weren't quite bright enough. So I put some tissue paper over the [NS-10M's] tweeters to tone down the brightness a bit.

R-e/p (SB): Do you use NS-10s exclusively, or do you also refer to the room monitors?

BC: I *never* rely on the room monitors; I've never heard any two pairs of studio monitors sound alike. So what do you get used to? You could drive yourself nuts. It's hard enough to get used to the way Yamahas sound in different rooms!

R-e/p (SB): You find that room acoustics make a difference even with close-field monitors?

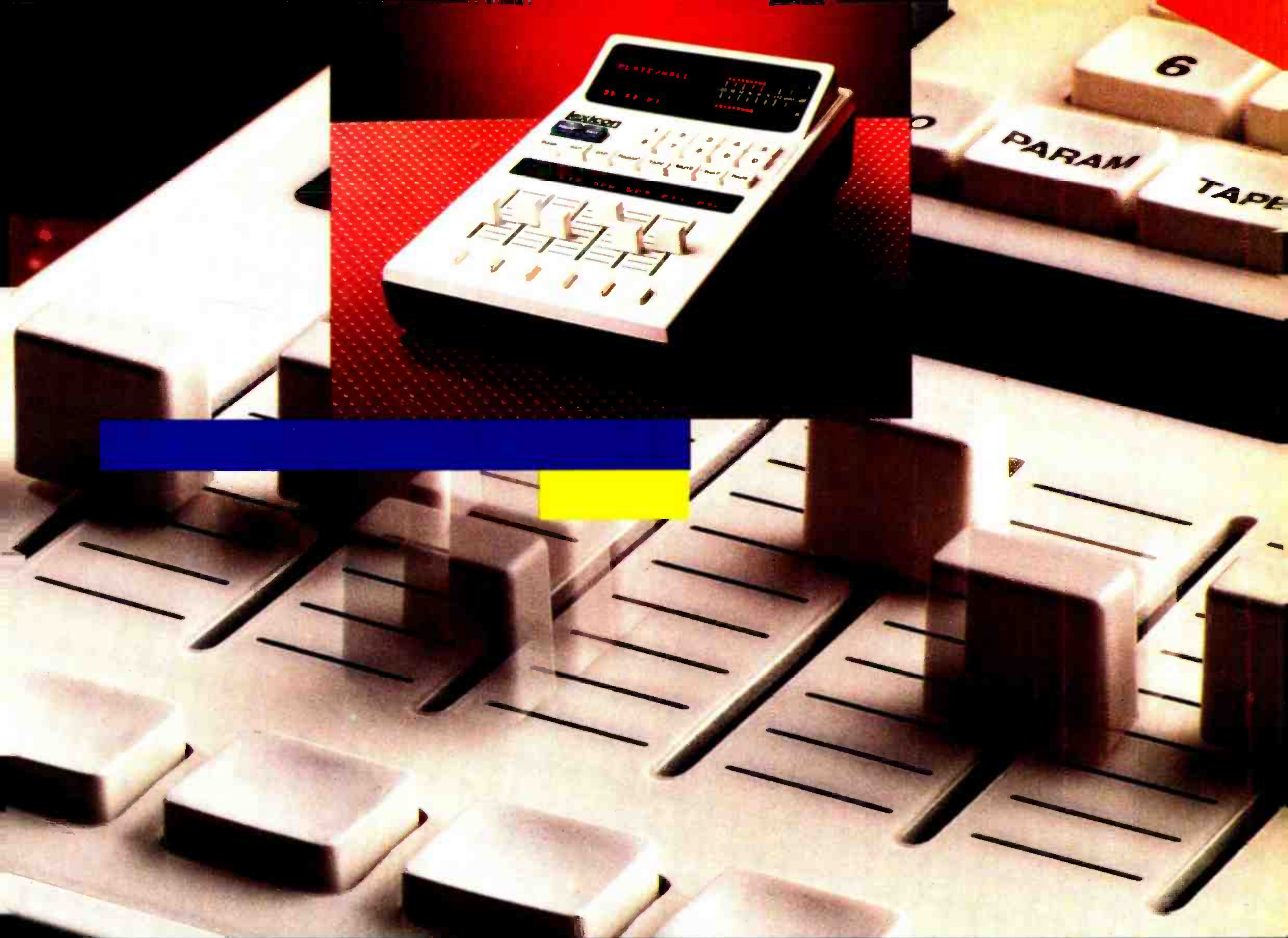
BC: Yes, and what really makes a difference is the console. They sound much different sitting over an SSL than over a Neve, for example. I'm not sure why — maybe it's the angle, or the mass. With the SSL, I usually set them back on a pedestal.

R-e/p (SB): Which rooms are your current favorites for mixing?

BC: I do most of my work in the SSL rooms at the Power Station, although I did the new Hall and Oates album in the new room at Electric Lady [New York].

When I first started at Power Station, they had a Neve 8068, non-automated with no grouping faders — no VCAs at all. I mixed a lot of records on that console, including *Flesh and Blood*, and a bunch of Chic records. I had used automation before that, at Media Sound, which had an API with Allison automation. I liked the concept of automation immediately, even though some of those early systems had problems. I talked the Power Station into getting one right away.

R-e/p (ML): Does the so-called "VCA Sound" bother you at all? Do you prefer



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to mix with a moving-fader servo system?
BC: No, I don't like the moving-fader approach. I see only one advantage to that technique, and that's when you bring an artist or producer into the studio who doesn't know about VCAs, and it lets him know immediately what is going on. There's an advantage to a VCA-based system that makes it much hipper than the moving fader. With the VCA design you can do a pass on the automation where, for example, you are riding the dynamics of a vocal because there are a few things you want to accent. Having done all of those little rides, on the next pass you can put your finger back on the fader and do an overall relative ride to, say, bring the first verse up a little bit, but all your previous dynamic rides are kept. You're just adding to what you did before. But with [NECAM], if I understand it correctly, unless you're grouping, as soon as you touch the fader, you override your previous rides. You can move it up and let go of it, but if there had been a previous ride at that point you don't get it in your mix.

R-e/p (SB): Concerning the sound of an automated console, do you not hear any of the VCA characteristics that engineers claim degrades the sonic clarity?
BC: I don't know, maybe I've just gotten used to it! I went from a console without VCAs to an SSL, and I prefer the sound of the SSL; I've never had a problem with it.

R-e/p (ML): Do you use the VCA input faders during tracking, or fader swap to the monitor faders?
BC: Generally, I'll run the mikes through the little faders to the multitrack. I suppose I do that just because people tell me the VCAs are doing something to the sound. But sometimes I have trouble understanding what people say about the sounds of the different boards. I've heard people say they don't like the sound of an SSL, that it has a cold, sterile sound. Well, I've been mixing on the SSL for years, and I don't have any problem with it. I think I know what people are missing, and that's all the transformers.

R-e/p (ML): Do you use the individual compressor/limiter and gating functions fitted to the SSL's input strip?
BC: Yes, quite a bit. Having one gate per channel is great, and they work very well. The company should make them as outboards for studios without SSL consoles.

R-e/p (SB): Do you use a lot of compression, limiting and gating to keep your mixes strong and punchy? Is that part of the "high-impact" Clearmountain sound?
BC: Well, yes and no. I try to approach each project differently, but there are some basic things that I do — certain kinds of echoes I like to use. For example, if I get a Huey Lewis tape, I might use the SSL compressors on the percussive instruments, to change the attack around quite a bit. But if they don't sound good, I'll turn them off; there's no one thing you can do in every case.

R-e/p (ML): What about other types of outboard gear: reverb, pitch shifters, and that sort of thing? Any favorites? Any general approaches for using them on a mix?
BC: One basic item for me is a stereo pitch shifter. I usually use the Publison

[DHM89B2] because it's stereo in and out, but I also use the AMS [DMX15-80] or two Eventide [Harmonizers]; it doesn't make much difference as long as they're clean. I'll send them off the stereo aux send, and set one side slightly above the pitch and one slightly below, both well within 1%. Then I'll just dump things in to see how they sound — on a guitar you'll get this slow flange kind of sound. Sometimes, if you have something on the left, and you want to expand it across the stereo sound field, you can add in some right-side Harmonizer. It's never enough to change the pitch; it's very subtle.

I'll also keep a stereo slap going, usually a tape slap. I'll get [the delay] so it's exactly in time, and is doing a quarter-note or eighth-note delay, or eighth-note triplet. I'll set it up by ear, usually with the click track. And, since it's stereo, if I use it on anything in mono that I select to the left and right, it will just be a mono delay. But if I take something from the left and put it into the right side, it will go right.

R-e/p (SB): I notice from listening to your mixes that, with percussion, you do a lot of spreading left and right.
BC: Yes, sometimes I'll do that with separate chambers. Maybe I'll have an instrument by itself with its own chamber, all the way to one side. Sometimes I like to give one instrument its own wacky echo off to one side, so it sounds as if it is off in another room, and you're listening through a doorway. A good example would be some effects on the Roxy Music album.

R-e/p (SB): Your drum sounds have always been particularly impressive. How do you develop these sounds from the recording session through the mix?
BC: First of all, I use my own snare drum. It's nothing special, just a Ludwig Black Beauty, 6½ inches deep which, unfortunately, they don't make anymore.

R-e/p (SB): So you just hand that to the drummer and say, "Play this one?"
BC: Yes, and I tune all the drums myself. Usually I'll ask the drummer *not* to touch the tuning at all, unless something gets loose. I try to keep lug locks on them so the drummer doesn't have to worry about that at all. If I have to, sometimes I'll even go out between takes to do a quick tune-up.

I'll try not to put anything on the drum heads, because I like drums to ring a little bit. If you tune a drum right, and the guy hits it right, you don't have to put any tape on it.

R-e/p (ML): What mikes do you normally use on the drums?
BC:[laughter] Do you really want to know that? I'm not sure it *really* makes all that much difference.

R-e/p (SB): Perhaps, but at least it gives R-e/p *readers a reference point to work from.*
BC: Well, I really think everybody should experiment and develop their own thing with mike technique. But basically, I use a lot of [Sennheiser] 421s: a 421 on the bass drum, and on the tom-toms, top and bottom. I usually use a Shure 57 and 81 on the snare, both on the top. Sometimes I'll use just one, sometimes both. I use AKG 452s on the cymbals, although I've just discovered the new AKG 460, which seems to be an improved version. It seems to have a

"When somebody brings in a tape I've never heard before, the first thing I do is push up the faders and get a rough mix . . . I'll solo things once in a while, but I'll listen to the vocal mostly, to get an overall attitude. Then I'll break it down and start building the mix."


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little more body to it, more depth. I'll usually stick up a couple [Neumann] 87s for room mikes, and aim them at the walls or the ceiling.

R-e/p (ML): How far away do you place the room mikes?
BC: It depends on the size of the room; maybe 10 feet away from the walls, facing the walls.

R-e/p (SB): How many tracks do you devote to drums, and how do you group them?
BC: Usually one for bass, one for snare, stereo drums including cymbals and tom-toms, and usually a hi-hat, unless I'm worried about running out of tracks. And then I'll have a pair of room tracks, usually compressed.

R-e/p (SB): Are you still working mostly 24-track, or have you moved up to 46-track sessions with work/slave tapes?
BC: I'll record on 24 and, if I'm running low, I'll make a slave and then mix on 46. I'm doing that more these days. I didn't do it at all with the Bryan Adams project I just finished, but with the Hall and Oates we did 46 with everything, because it was full of tracks and tracks of synthesizers and drum machines. [See attached track sheet for "Some Things are Better Left Unsaid" — *Editor.*] There was one song — which didn't end up on the album — that had a full set of drums, a full set of Simmons . . . and Linn Drums forever! We had a 24-track tape filled with just drums!

R-e/p (SB): I'm curious about one snare drum sound in particular: the title track of Springsteen's Born in the USA. *It has the impact of a .38-caliber revolver going off. How did you capture that?*
BC: What I did for a few songs on that album — and I think that was one of them — was to use the great sound I'd got from the stereo overhead mikes. The snare sound was amazing, for one thing because Max [Weinberg] tunes his drums really well. The snare drum mike itself wasn't happening, maybe because it was too close, but the overhead mikes were picking up this "Glyn Johns" kind of snare sound. So I just sampled that into an AMS, and it became the predominant snare drum sound, although it is mixed in with the original snare drum track. It was easy to do because there are no other drums playing during the intro part.

R-e/p (ML): So you're triggering that sound out of the AMS DMX15-80 for each snare beat?
BC: Yes, any signal you feed in will key it. You can also put little vocal snippets into the AMS, and key it off something like a bass drum. On the intro of the Hall and Oates album, there are some vocal bits, singing some Spanish words. By keying one off the bass drum and one off the snare, we have these little vocals answering each other exactly in time with the Linn Drum.

R-e/p (SB): Do you have any general rules for setting up a drum mix?
BC: Not really; I just try to get them to sound as punchy as possible. The only basic rule is that there's a certain balance between the bass drum and the snare drum I always go for, and also a certain balance between the bass guitar and the bass drum. The blend of those sounds is very important.

R-e/p (SB): Are you conscious of arranging the instruments across the stereo soundstage, particularly with regard to the mood of a song? On the Springsteen album, for example, the drums are centered and fairly dominant in most songs, but on "I'm on Fire" they are panned to one side and almost sound like they're in another room.
BC: Yes, although that song was an exception. Most of the album is set up so it's like watching Bruce while he's playing on stage. If your stereo is wired correctly, the organ is on the left, and the piano is on the right, lead guitar is off to the right a bit, because that's how they set up. But on moody songs like "I'm on Fire," I'll put the drums outside, and have the spotlight on Bruce as if he was standing alone with his guitar.

R-e/p (SB): Do you set up your mixes while working alone, without the artist or producer in the room?
BC: Yes, usually. Sometimes I will do complete mixes by myself. I've done some [mixes for] Australian bands where they send the tapes and stay down there. But usually I will set up the mix alone . . . Bruce would give me two or three hours to set up before he would come in.

R-e/p (ML): What kind of input do you like from an artist or producer? When he or she comes in after you've set up your basic mix, what do you need to hear from them to help you finish up to everybody's satisfaction?
BC: [laughing] I'd like him to tell me it sounds perfect just the way it is!

R-e/p (ML): Let's assume they don't.
BC: I'd like them to say things like "The vocal should be more present or more up-front," or "The drums are too understated" — those kind of things. The obvious thing would be to say, "The vocal needs more echo, so why don't you put a 7½ slap on it?" I'd rather they didn't do that; I'd rather they give me a description of the mood they want, because that leaves me some room to visualize it in my head and then go for it; possibly something more unusual.

R-e/p (ML): Do you prefer to keep the producer away from the console?
BC: Usually, but once in a while somebody will have a specific move they want to do. I was working with Pat Metheny the other day and he heard something that I wasn't quite getting. I asked him to give it a shot and he got it right away. That's the great thing about computer automation.

But getting back to taking input from the artist on the mood, I was working with the Divynls on their record and after about three days they sat me down and said: "You're missing what we are all about. You're making the mixes too big, too broad, too high fidelity." I had just finished mixing Roxy Music, and I was into Bryan Ferry's big, wide, roomy sound with lots of delays; they told me they weren't into that. They wanted their records to sound *small*, like a little spiky ball with sharp edges sticking out of it. I immediately understood what they meant.

I went back, listened again, and realized they were absolutely right! I was working so hard to make records sound big that I didn't think of going in the other direction. From then on I tuned

BOB CLEARMOUNTAIN

right into it: they wanted it to sound jagged, rough, and closed-in. I learned a lot from doing that. Sometimes records *shouldn't* sound big.

R-e/p (SB): Let's take a small detour here and talk about digital technology. I notice that the only album you've remixed digitally was the Bryan Adams' Cuts Like a Knife. *Why did you mix to digital on that album?*
BC: Very simple. Bryan was recovering from some surgery after we finished the tracks, so instead of mixing at Power Station we went to Le Studio in Canada, because he didn't want to be in New York. They didn't have a half-inch two-track up there; instead they have a JVC [BP900] digital processor. Right after I did that, I mixed the Bowie *Let's Dance* album, and there were some things about the top-end on that which I preferred to Bryan's record.

R-e/p (SB): The Bowie album was mixed to half-inch analog?
BC: Yes, although there were other differences. The Bowie project was done entirely on the SSL, where Bryan's was recorded on a Neve and mixed on the SSL. The first thing I thought of [to explain the difference in sound quality] was the digital. But then the Compact Disc of Bryan's album sounds incredible; it sounded better than I ever thought the record had sounded, even when we were making it!

R-e/p (ML): Have you recorded any projects to digital multitrack?
BC: No, none at all, so I don't really have any opinion about it. My favorite medium is the one that is most reliable. Digital scares me a little bit, because I've heard stories about things just not coming back. So I'm just waiting for it to be perfected; waiting for a standard to be agreed upon.

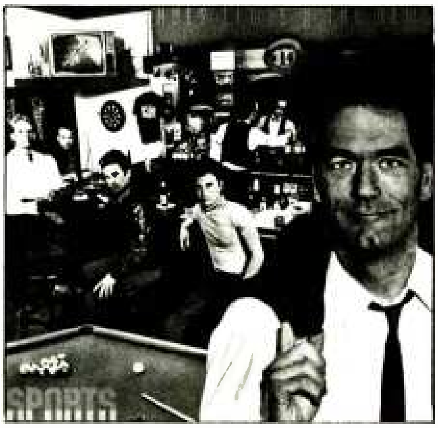
R-e/p (ML): If the reliability factor for digital was the same as analog, would having 32 available tracks be an advantage?
BC: Probably, if I could be sure that it would work, and if I could edit on it, either electronically or by cutting the tape. I'd actually prefer electronic editing, but I can't see spending all that money for two machines when I can cut analog tape.

R-e/p (ML): Do you do a lot of tape cutting on analog multitrack?
BC: It depends on the project. On Bryan Adams we did a lot; on Hall and Oates not quite as much, although we did have to cut some parts out just to shorten the songs.

R-e/p (ML): And I assume you edit on two-track after the mix?
BC: Sometimes. For the Rolling Stones *Tattoo You* album, because what we had was way too long, we ended up doing a *lot* of editing after the mix. The record was 26 or 27 minutes each side when we finished it; there were *11* songs on it. Mick said, "We've got to make it shorter," and I said, "Well, let's take off a song or two." He didn't want to do that, so we went in for about a week and a half, just cutting little pieces out. Mick would say, "Make this one shorter." So I'd go in and take some bits out, and say, "Do you want to listen to the edit?" "How much did you take out?" "About thirty seconds." "That's not enough. Go back and take out a minute."

R-e/p (ML): Did you dub the mix first, and edit on the copy?
BC: No, this was the original master tape, but I kept all the pieces, all carefully logged.

R-e/p (ML): Did you have to put some of the pieces back in?
BC: Yes, a few!



R-e/p (SB): I understand that you've been working with Mick Jagger recently. Can you say anything about that project?
BC: Yes, he's doing a solo album, with a couple different people producing. The ones I'm involved with have Nile Rodgers producing. I've only heard the one song I mixed, and it was fantastic. It was interesting to hear Mick singing with musicians other than with the Stones, although the particular track I mixed does have a "Stones-type" feel.

R-e/p (SB): Who are the musicians playing on the project?
BC: A drummer named Anton Fig; Bernard Edwards from Chic on bass; G.E. Smith; Rob Sabino, who also was with Chic, on keyboards — a lot of Nile's guys.

R-e/p (SB): I notice that you are working more and more into the role of co-producer. How do you function when you're wearing another hat . . . with Hall and Oates, for example?
BC: Well, with Hall and Oates I'm more of a glorified engineer, because they really produce themselves. They take suggestions from a lot of people, like Arthur Baker and myself. Actually T-Bone Wolk, the bass player, deserves production credit as much as anybody because he did most of the arranging. My producing role is more a function of what I do with the sounds; the way I work them into the record.

A good example might be a song called "Some Things Are Better Left Unsaid," which is based on Daryl's guitar part, and he wanted to have some kind of interesting sound for it. I set up a hi-hat pattern with eighth notes on the Linn, then put his guitar through a gate and keyed it off the hi-hat. It pretty much turned the feel around, and he got right into it. It was probably the feel he had in mind, but he needed some help in getting it. It's things like that, where I'm not actually telling people to do this or do that.

But sometimes I'll do something with the sound that will give them an idea to do something else. I took all the ideas and made them work together, but all good engineers and good mixers do that.

R-e/p (ML): Maybe, but it can be a difficult role sometimes. Engineers can get frozen in the spotlight, so to speak, and don't want to take the responsibility. A good engineer has to step out sometimes to make things happen?
BC: Right. And I think Hall and Oates enjoyed working with me because, no matter what the situation was, I would never say, "No, that can't be done." I would always go after it. Sometimes it couldn't be done *exactly* the way they wanted it, but maybe I could get something close; there's always some way to get the idea on tape.

R-e/p (SB): Right now you're mixing for at least a half dozen artists, including Bruce Springsteen and Mick Jagger. How did you end up in this enviable position?
BC: I think they must pick up on work I'd done before. With the Stones, I'd done a mix for them on "Miss You," which originally was supposed to be just a dance mix. At that time I was doing a lot of dance mixes, and back then it was unusual for a rock band to want one. Because I had worked with Chic and Narada Michael Walden and other Atlantic projects — and Rolling Stones Records had distribution through Atlantic — the head of Rolling Stones Records, Earl McGrath, suggested they give it to me for the mix. They liked the dance mix so much they had me do the single.

R-e/p (SB): And Springsteen?
BC: That story goes back aways. I did an Ian Hunter album when the Power Station first opened; he was one of their first clients. Ian was friends with some people in a band I'd produced called Tuff Darts. He came in and played on some Tuff Darts tracks. He liked the studio, even though it had just opened and it was unfinished — there was no door, the floor was bare concrete, the walls were sheet rock because the wood hadn't been put up! He liked the feel of the studio,

and I guess he liked what I did, because he came in to do his album, *You're Never Alone with a Schizophrenic*. Ian used the E Street Band on that project, and they loved the studio as well. The band went back to Bruce and said, "You should come over to this studio and check it out." He came over one night, did a couple tracks, and he loved it — Bruce recorded *The River* at Power Station. I worked with him for a couple weeks, but a conflict with a production project I was doing meant that another engineer, Neil Dorfsman, had to take it over to finish the album.

R-e/p (ML): Did you do any of the mixing on The River?

BC: I did about 10 or 12 mixes for Bruce in the middle of the time they were recording — they recorded for about a year — and he kept one of them, "Hungry Heart." So we've known each other since then, and I guess he liked what I did. Actually, he heard Steve Van Zandt's album, which I had just finished right before he was ready to mix *Born in the USA*, and I guess he liked that, too. So he called me up, I mixed three sides, he listened to them and asked me to finish the album. That was a *very* well recorded record; Toby Scott did an excellent job.

R-e/p (ML): From your list of production credits, I see that you have experience with mixing for four somewhat different mediums: records, radio, video, and film. I'd like to talk about the technical side of the Stones movie, Let's Spend the Night Together, *which obviously meant you were dealing with something very different from an album mix.*

BC: The Stones movie was actually a lot of fun once the basic editing was done to conform the multitrack [tapes recorded live on location] to picture; that was done brilliantly by Bill Marino at Regent. [The interested reader is referred to Steve Barnett's excellent two-part article describing the application of multitrack audio production during the re-recording of the Stones movie, published in the December 1982 and February 1983 issues of *R-e/p — Editor.*]

Some of the edits were very tricky, because they did some crazy things. There are a couple places where the picture edit would turn the beat around. And there was one place where I had to repeat a section because half the band went to a bridge and the other half didn't, and then a bar later they all came together. I did a little offset with the [Audio Kinetics] Q. Lock, repeated the instrumental tracks, and took out the vocal.

R-e/p (ML): If I remember correctly, you prepared a multitrack "premix" that was used during re-recording of the film's soundtrack at Todd-AO in Hollywood.

BC: Basically I gave them two options. I was mixing from one 24-track to another 24-track, so I was able to have some flexibility. I gave [the re-recording engineers] five "behind-the-screen" tracks, because I couldn't get any perspective on the surround channel. Then I filled the other channels on the multitrack with individual instruments and vocals, all post-fader and post-EQ. So if they wanted to, they could turn down the five-channel mix, put all the other faders in a straight line, and they would have an adjustable, "automated" mix. The echoes were separate, and they could change speaker assignment if they wanted to. They told me they basically kept my five-channel mix and occasionally would add something, a little more vocal usually. [The six-track mix of *Let's Spend the Night Together* was re-recorded in the original "discrete" six-track Todd-AO format, which employs five full-range channels behind the screen. Currently, most six-track film mixes are mixed in the Dolby "baby boom" format, which only has left/center/right full-range channels; left-and right-center channels contain only bass information below 200 Hz — *Editor.*]

R-e/p (ML): You didn't go to the final re-recording session at Todd-AO?

BC: I couldn't, because I was producing a second Garland Jeffreys album at the same time. Actually, to start with I didn't even know I was going to be working on the movie. I was mixing the live album [*Still Life*] when Mick Jagger casually mentioned, "Well, when you do the mix for the movie . . ." and I go, "What?" because I already had all these projects lined up for the rest of the year. But the fact that I'd done the live LP helped the movie mix go quicker, because I was familiar with all the songs. Of course mixing with five speakers behind the console is different than mixing with two. It sounds so cool, sitting behind these five Yamaha [NS-10] speakers, and there are so many different things you can do.

R-e/p (ML): Let's move on to another audio-visual medium, the HBO TV specials you've mixed for David Bowie and Hall and Oates. Did you mix those in stereo?

BC: Yes, because usually there would be an FM simulcast with the first showing. I mix those pretty much like a live album, although I'd watch the picture to accent certain things that appear on the screen — hopefully you have a final video edit to work from. Sometimes it's difficult if they're going to re-edit, as they did with the Bowie show. But with those projects the most important mix is the mono, so I brought in an old, cheap TV that was sitting around the Power Station, and ran the audio through that, after squashing it through an LA-2A [limiter] to simulate what [the TV stations] will do to it. I work to keep the stereo as compatible with the mono as possible, by not panning anything extremely and watching phase very carefully.

. . . continued overleaf —

"When you're overdubbing, you're focused in on a specific performance — with the mix you want to step back and listen to the *whole* record."

But when it comes on TV it's gone through who knows how many compressors, and is usually noisier than hell because it's coming off a piece of videotape that was not made to record audio in the first place.

The Hall and Oates special wasn't bad but on Bowie's show they added applause and a lot of echo; I don't know why they did that, because there was plenty of echo to begin with. The Hall and Oates project was actually a remix, and I think one of the reasons Daryl didn't like the original mix was because of all the sleazy applause tracks added in later.

I enjoy doing them, but it scares me because I don't think I've done one yet where it came out the way I thought it should.

R-e/p (SB): I assume that you have more control with record projects because you work very closely with mastering engineer Bob Ludwig at Masterdisk. Will you lose that kind of control with Compact Disc mastering?

BC: No, because the final Sony 1610 masters used for the CD are done by Bob at Masterdisk. So anything we've done for the record is done for the CD, unless we decide differently. Bob is *very* conscientious about that. In fact, he was telling me that when he makes the digital master, he will sit there and watch the meters through the whole album once to see what kind of headroom he has, so that the loudest point in the record just touches the limit. That way he gets as much dynamic range on the CD as he can.

R-e/p (SB): Of course, you could get into trouble while mixing for "super-fi" home systems with that kind of dynamic range, because most people who listen to your mixes don't have them. Do you check your mixes on more conventional playback systems — Walkman-type portables, boom boxes, car stereos?

BC: Yes, I try to. On this last Bryan Adams project, we had a couple of those little Aiwa speakers you use with a Walkman. We would run the mix onto a cassette, stick it in the Walkman, listen to it, and listen on headphones. The only thing I usually *don't* check it on is the big studio monitors, although every so often we'll turn them on just for a laugh!

R-e/p (ML): As Music Videos gain in popularity, some video directors are going back to the multitrack to add effects and dialog, and then remix for that medium. Is that an area that interests you?

BC: Sure. I'd love to be involved in it, though the film-sound mixers and effects guys have the experience, and already know all the tricks of the trade.

R-e/p (SB): But the video soundtracks of the songs you have mixed are basically just straight dubs of your record mix?

BC: Yes, up to now. But tomorrow night I'll be doing a mix of Hall and Oates' "Method of Modern Love" just for the video, because the director decided he wanted to accent something that happens in the track — it's just one small vocal move; the rest of the mix will be the same. Since it's all on the SSL with Total Recall, we'll just put on the floppy disk and it should all come out the same. We hope. There are problems with that — not with the SSL, but with outboard equipment.

R-e/p (SB): What kind of problems?

BC: Because I'm doing all my mixing with SSL automation, I'm coming back to "fine tune" a lot of mixes. On the Bryan Adams album, for example, we remixed everything on the album at least once, and one song we remixed six times in three different SSL rooms, using Total Recall. The Total Recall system works amazingly well, especially if you're using the same specific board; if you're using a different board in another studio, there can be slight differences but it usually works fine.

The big differences come from the outboard equipment, and that's what can really get into trouble trying to reset levels to match a previous mix. So many of these companies don't have true unity gain settings. One digital delay has an input level on the front, and the output level is on the back. What are the manufacturers thinking? "Aha, this will get them! Let's put this on the back where they can't get to it." Down at Electric Lady I spent half the time crouched behind the rack, trying to make the input and output levels right. But who *knows* where they should be in relation to each other? They should have a unity gain switch, or an input and output control inversely proportional, so you can get the right level on your headroom meter, but still have the same level coming out.

R-e/p (ML): Why would you need input and output controls at all?

BC: You might if you want to optimize the signal-to-noise of that unit, or to add gain at the input.

R-e/p (ML): But don't most consoles have master send controls to let you trim the level to the outboard unit?

BC: Right. All you really need is a +4 in and +4 out. So why do they put these ambiguous level controls on? One DDL I use regularly is a $10,000 device, with remarkable sound quality, and everything about it is fantastic — except for the inputs and outputs. It's a stereo device, and both outputs are ganged to one pot — two separate input levels but only *one* output level. It's crazy! They have numbers 1 to 8; what does that mean? I find that +4 is somewhere around 7. But then where do you set the output level? What's optimum? Why do they do this to us? How do you come back to a mix? You have to mark down that the knob was at "7-plus-a-little-bit." If they're going to have these controls, at least give us a switch to give us +4 in and +4 out!

With the complexity of the mixes I'm doing now, it can be very annoying dealing with these things time after time. With the outboard level problem, we have to put tone through every piece of outboard gear and line it up before we start, which takes an extra half hour.

R-e/p (SB): With all these new complexities, do you sometimes long for the simpler days, back when you first started in this industry?

RP: It sure was different! When I was working at Media Sound, I used to do projects with people like Al Martino and Englebert Humperdink. I'd line up all the faders, put the vocal about 20 dB louder, and add tons of echo — each song would be the same. When you got to the next song, you wouldn't even break down the board because the tracks were all in the same places. Times certainly have changed since those days.

R-e/p (SB): Any plans for the immediate future?

BC: Yes, a vacation — even if only a short one. A day off every year or so would certainly be nice! ■■■

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SOUND-SYSTEM INSTALLATION FOR THE SUMMER OLYMPIC GAMES

Panasonic RAMSA Sound-Reinforcement Systems Installed and Operated by dB Sound

by David Scheirman

Sound-reinforcement systems come in all shapes and sizes. When an Event Coordinator casts a roving eye on the wide variety of audio hardware available on today's sound reinforcement scene, it can look like a jungle out there. What gear should be specified? How does it get to the site? Who will operate it?

The 1984 Summer Olympic Games, held at various venues within the greater Los Angeles area during July and August of last year, presented a formidable challenge to the sound reinforcement community: International broadcast feeds, a plethora of portable spectator and press-interview sound systems, and a new, permanently-installed sound system for the 90,000-seat Los Angeles Memorial Coliseum were all part of the equation.

In an effort to obtain a single-source supplier for all its sound reinforcement needs, the Los Angeles Olympic Organizing Committee put forth a request for bids to various sound-system manufacturers. The Los Angeles-based consulting firm of Smith, Fause & Associates was hired to survey the various sound system requirements, and assemble bid specifications. Nearly a year before the Olympic Games actually took place, Panasonic Industrial Company's Professional Audio Systems Division had been chosen to supply its RAMSA product line.

"Perhaps the single greatest concern of the LAOOC," explains Tom Bensen, marketing manager for Panasonic's RAMSA New Technology Product Group, was that a broad-based equipment supplier with a proven track record be named the single-source supplier for sound reinforcement needs. The RAMSA gear had proven itself in heavy-duty use for special events such as the 1982 World's Fair. And the company's ability to supply everything from microphones to loudspeakers, including the super-directional horns necessary for the Coliseum sound system, gave us an edge."

Even the largest manufacturing firms often do not maintain staff personnel to set up and operate their product lines in actual-use situations. dB Sound, Inc. of Des Plaines, Illinois, was chosen by Panasonic as the subcontractor best suited for system packaging, installation and operations. Perhaps best known as a supplier and operator of high-level sound reinforcement systems for touring use, dB Sound was able to draw upon such resources as an in-house case and cabinet manufacturing company, and access to a wide variety of support equipment lines through a separate division for retail sales, known as Music Dealer Service.

"We have had a tremendous amount of experience dealing with outdoor festivals and multiple-stage setups,"

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SOUND FOR THE OLYMPICS

confides dB sound owner/partner, Bruce Gordon. "We were very much aware of the potential problems associated with setting up and operating a group of sound systems in many different environments, with a tight schedule to work with. Our expertise in this area made us a good choice to deal with the actual setup and operations of the systems. As it turned out though, the advance packaging and system preparation took much more time and energy than actually servicing the events."

To keep in touch with the rapidly changing system specifications prior to the Games, dB Sound set up a 3,500 square-foot office and shop in centrally located Hawthorne, California. Jim Ash was appointed director of the new dB/West operation in the fall of 1983.

"In the beginning, things were very vague, just ideas with no form," Ash notes. "On the second Wednesday of every month, starting in July '83, there was a status meeting during which the LAOOC would take a look at how the 'final plan' was doing. The only problem was that the 'final plan' kept changing! We realized early on that if this whole project was run from the corporate level, it could be in trouble when the deadline came around. So we decided to take the ball and run with it."

dB Sound assumed responsibility for the assembly and operation of 39 temporary sound systems, for use in at least 24 different venues during the Summer Olympics. Spectator sound systems were required wherever public groups were assembled to watch the various competitions, plus athlete call-up systems for use in paging competing athletes from the practice areas. In addition, 26 press-interview systems were left set up in each press room at the different sites.

A total of 930 speakers, 250 microphones, 215 amplifiers and 57 mixing consoles were included on RAMSA's equipment supply list for these systems. In addition, ancillary gear such as cassette decks, turntables and equalizers were provided. The bulk of this latter equipment came from Panasonic's Technics Division.

The system-use period stretched from July 27 to August 12, when final closing ceremonies were held at Los Angeles Memorial Coliseum. Event sites stretched from Malibu Beach, northwest of Los Angeles, to Fairbanks Ranch in San Diego County.

"Once the two main installed systems [at the Los Angeles Memorial Coliseum and East L.A. College] were in and operating, and the smaller portable system packages put together, the project became an exercise in logistics," notes dB's Jim Ash. "With everything pretty well set up ahead of time on paper, we were able to determine what our transportation, communications and personnel needs were. Each system and each site had its own special needs. It was up to us to assemble equipment racks with patchbays to suit each system."

Figure 1: dB Sound crew installing Panasonic WU-566/2 driver on WU-5948 long throw horn for L.A. Coliseum main speaker cluster.

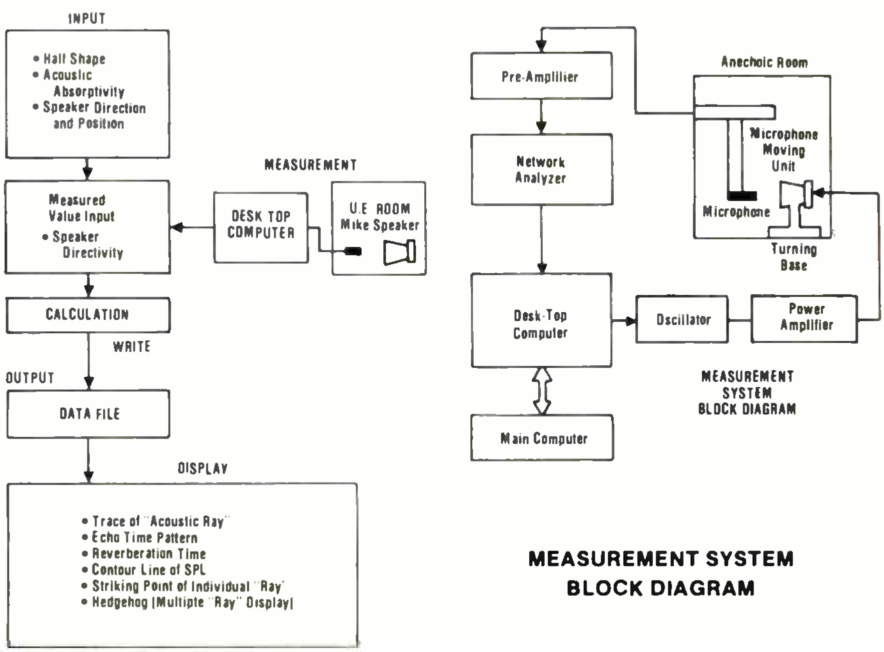
Sound System for Los Angeles Memorial Coliseum

By far the most complex sound-reinforcement system used during the 1984 Summer Olympics was the dual-source, permanently-installed cluster system designed for the L.A. Memorial Coliseum. Designed with the help of a VAXII/780 main-frame computer manufactured by Digital Electric Corporation, the Coliseum system featured RAMSA's new WU-S948 super-directional long-throw horn (Figure 1).

The concept of hall computer simulation developed by Acoustic Research Laboratories of Matsushita Electric Industrial Co., Ltd., is based on an "acoustic ray" method, which reportedly offers the advance calculation of sound-source effects on acoustical spaces. Input data may be weighted according to the directivity factor of various loudspeakers, as shown in the block diagram, Figure 2.

Chief system engineer Kimio Takei, of Matsushita's Audio/Video Division, supervised the installation and setup of the Coliseum system, which featured 16 WU-S948P horns, 12 WU-S968P twin-Bessel horns (Figure 3), and 24 RAMSA WU-S907/8WP 15-inch waterproofed bass loudspeakers. Twin loudspeaker housings, measur-

Figure 2: Block Diagram of Acoustic Ray Method to Calculate Overall Directivity and Coverage of Sound System Components Specified for Summer Olympics. (Taken from "Technical Report of Sound Pressure Distribution by Computer Simulation for the Los Angeles Coliseum," by Katsuaki Satoh, chief engineer, Acoustic Research Labs, Matsushita Electronics Industrial Co. Ltd.)



MEASUREMENT SYSTEM BLOCK DIAGRAM

ing 15- by 15-foot contained the loudspeaker arrays, which were powered by 30 RAMSA WP-9210 dual 200W amplifiers (Figure 4). Model WZ-9320 equalizers and WZ-9420 electronic crossovers also were installed in the system.

dB Sound's Jim Matheson headed the installation team for the Coliseum. Overall system performance parameters were met, the engineer reports, including the requirement that a sound pressure level of at least 91 dB be measurable at the back rim of the Coliseum, approximately 1,200 feet from the sound source.

The system's front-end includes a RAMSA WR0-8616 mixing console. One of this product's unique features is a six-channel remote start/stop control section for program materials using turntables or tape cartridge machines.

(One important note: the LAOOC let the audio services contract for Opening and Closing Ceremonies at Los Angeles Memorial Coliseum to Estrin Associates, Inc. Subcontractors Best Audio and Stanal Sound provided equipment and technicians for just these two events.)

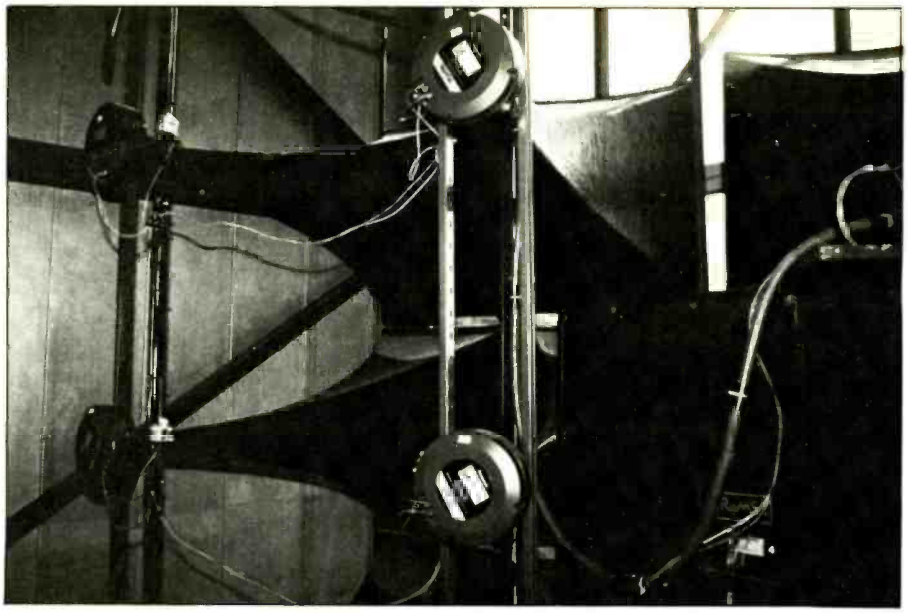


Figure 3: The Coliseum system comprised 16 Panasonic WU-S948P horns, 12 WU-S968P twin-Bessel horns, and 24 RAMSA WU-S907/8WP 15-inch waterproofed loudspeakers.

East Los Angeles College

A second, permanently-installed sound system that was left behind once the Olympic Games had concluded is at East L.A. College's outdoor stadium (Figure 5). Here, twin speaker towers flank the large scoreboard at Weingart Stadium.

"At first, we thought there would be a neighborhood noise level problem," Ash recalls. "The stadium is very close to the adjacent residential area. And the speaker system that we put in there is capable of producing truly 'rock-and-roll' sound pressure levels. It serves only 25,000 persons, compared to the L.A. Coliseum's 90,000 seats, yet it has approximately two-thirds the number of bass transducers.

"That system is going to serve the college well for years to come, no matter what type of program material they throw at it. As it turns out, many of the area residents actually enjoy being able to hear the college's athletic events, which are all held during the daytime hours. So that system's long-throw capabilities do not present a problem."

The East L.A. College system features a RAMSA WR-8112 mixing console that is housed in a control room overlooking the playing field (Figure 6).

Santa Anita Race Track

The only hanging system to be used at any Olympic Games site was suspended from a steel basket at the Santa Anita Race Track . . . and then not used!

"We flew the cluster at a central point in that facility," Ash continues. "Right from the start, it turned out to be a problem because the broadcast commentators were positioned directly underneath the speaker platform. This meant, of course, that there was a bleed-through of spectator-area announcements into the on-the-air microphones. That was not acceptable.

"At first we tried re-equalizing the system, and turning down most of the low-frequency components. Ultimately, a set of separate perimeter speaker units was required to solve the problem."

Equestrian events were held at this world-class thoroughbred horse racing facility. The temporary sound reinforcement system added for the Olympic events serviced spectator crowds numbering in excess of 50,000.

Figure 4 ***(left)*****: Power for the L.A. Memorial Coliseum loudspeaker arrays was provided by 30 RAMSA WP-9210 dual 200W amplifiers.**

Figure 5: ***(right)*****: At the East Los Angeles College towers were located either side of the Weingart Stadium's large scoreboard.**

SOUND FOR THE OLYMPICS

Figure 6: Sound mixing duties at the East L.A. College were handled by a RAMSA WR-8112 console located in a control room that overlooked the playing field.

University of Southern California

A spectator area holding 11,000 persons for the new Olympic swimming pool and diving well, built on the USC Campus for the 1984 Games, was served with a temporary system (Figure 7).

"The synchronized swimming event required some real problem-solving," Ash says. "Kenny Landis actually had to do a lot of jumping into the water himself when that system was being set up, since digital delay units were required to compensate for the distance between the spectator speaker system and the pool, for instance. And then you take into account the fact that sound travels faster underwater than it does in the open air . . . it took a *lot* of pre-event adjusting to get that one right. We even brought in a synchronized swimming team to make sure that it would be right for the actual competition."

Separate portable systems were provided for the swimming and diving areas, and comprised RAMSA WS-100 loudspeaker units perched on folding tripods (Figure 8).

Fairbanks Ranch

Perhaps the most difficult system to set up in a short amount of time served the equestrian endurance events, held at Fairbanks Ranch in San Diego County. Here, a delay in grounds preparation on the far-flung golf course cut a scheduled three-day setup time to less than 24 hours.

"An amazing amount of preplanning was done down there," Ash confides. "The horses never actually got onto the fairways, so there wasn't much sod damage. However, the various crowd locations required a large number of separate loudspeaker horns mounted on poles, all strung together on a 70-volt [distribution] line."

dB project manager Bruce Gordon recalls that the uneven terrain prevented the use of the planned pickup trucks for system installation. "When they finally let us in there, we had so little time left that we commandeered two front-loaders with balloon tires, and worked with two, three-man crews, racing to get all of the 80 paging horns up. We had to siphon fuel, hot-wire the things . . . you name it. The system was put up and taken down in time, and we didn't go over budget. It took miles of cable, though!"

Lake Casitas

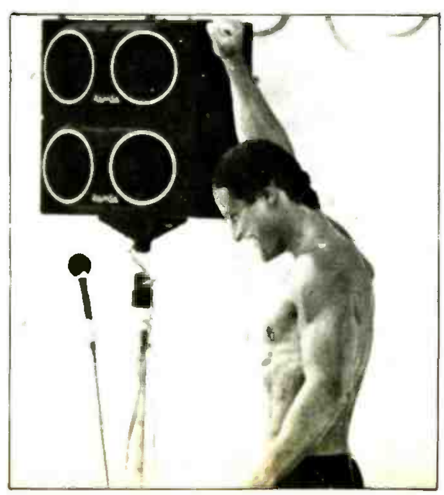
Rowing and canoeing events were held at Lake Casitas in Ventura County, 80 miles northwest of Los Angeles, where the approximate seating capacity in the spectator area totalled 10,000 persons (Figure 9).

"Here, we had some unique problems to solve," says Ash. "We had a separate system for the spectators, and one for the athletes' call-up area. The athletes wanted to hear the general commentary, in addition to being able to receive paging calls. You also had a system at the judging tower and the finish tower . . . and a 2,000 kilometer distance between the two!"

A special transmitter designed specifically for the events by Motorola was carried aboard the ABC network boat, and the commentators' dialog

Figure 7: Temporary sound mixing for the swimming competitions held at the new USC pool and diving well comprised a pair of RAMSA consoles and outboard equipment.

Figure 8: Olympics gold-medal winner Greg Louganis catches results at the USC pool beside WS-100 tripod-mounted loudspeaker.



SOUND FOR THE OLYMPICS

offered as an input signal to the athletes' sound system via a receiver. "Provision was also made for the boat to be able to flip a switch," Ash recalls, "and be in communication with the judging tower over their walkietalkies, then flip back to broadcast mode. The sound reinforcement systems only heard the actual commentary."

El Dorado Park

Archery events were held in El Dorado East Regional Park in Long Beach, an 800-acre recreational area that includes an archery range built in 1972, and the only one of its kind in Southern California.

"This was one example of our crews having to take the blueprinted plans of a site and making on-site decisions of changes," Ash notes. "The drawings that we had depicted speaker locations at three different ranges ... 30-meter lines, and so on. When we actually got there, it turned out that what they needed *were* three separate system operations, but *not* different systems on the different ranges. They needed to address the competing archers at the firing line, address the spectators, and also do an athlete call to the ready area. And the competition area was divided into men's and women's groups.

"One console was able to address various speaker zones: panpot to the left, you were talking to the men's area; panpot to the right, you addressed the women; and the monitor outputs bus accessed the 8,000-seat spectator area.

"This was just one of many instances where we were able to cut down on the number of actual system components required by doing a little signal routing. But it usually evened out, because other sites sometimes required more gear than was bargained for originally."

Long Beach Arena

Some event sites, including the Long Beach Arena, were already well-equipped with permanent sound systems and operating personnel. "For these sites, not much was necessary but to drop off the press feed system, which included microphones, WS-11U speakers on tripods, a WP-9210 amplifier, and a WR-130 mixer, and then check occasionally to make sure that everything was functioning correctly. Credit should really go to Joel Marx of the LAOOC's Audio-visual division, who kept track of a tremendous amount of the smaller gear at many venues," Ash confides.

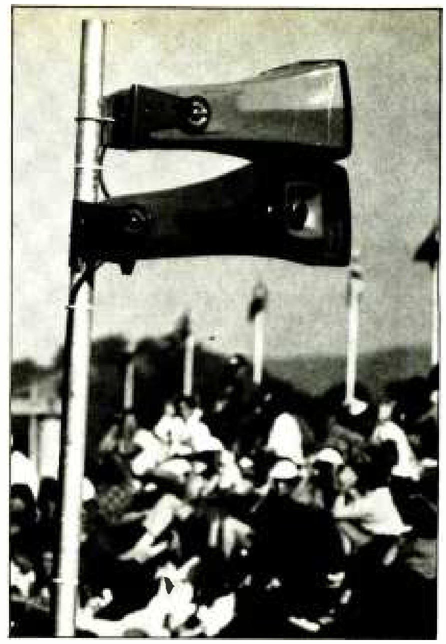


Figure 9: Panasonic long-throw horns were used at Lake Casitas to cover the 10,000-capacity spectator area for canoeing and events.

Pepperdine University

Water Polo events held at Pepperdine University's Raleigh Runnels Memorial Pool were viewed by 5,000 persons in an outdoor setting, where a roof-mounted loudspeaker system addressed the site (Figure 10).

"At this site, we were asked to allow the commentators himself to run the sound reinforcement system also," explains Ash. "The television people did not want to see another technician on-camera because of limited space at the console area. This did happen occasionally ... but, as a general rule, commentators are well-versed in audio operating techniques. And we had a man only 10 feet away from the controls at all times that the system was in use."

And Afterwards ... Cable Mountain

What do you do with the miles of Connectronics and Belden microphone and speaker cable left over from a series of events such as the Olympic Games?

"All told, we probably used 25,000 feet of Belden mike cable, and 20,000 feet of 12-gauge zip cord," Ash recalls. "Just in the various types of speaker cabling, we ended up with over nine miles' worth.

"As each system was taken down for the last time, it was all brought back to the shop. System components were boxed up and loaded into a semi-trailer for shipment back to RAMSA. All of the cable, though, went into a huge pile to await cleaning and sort-

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ing out after the events. The shop crew started calling it 'Cable Mountain' once the pile became about shoulder-high."

Some system components were sold to regional customers after the Games, Ash says. "Several of the school audio-visual departments were sufficiently impressed with the RAMSA mixers that we have seen additional retail sales as a result of using the temporary sound systems at those sites. However, probably the most positive aspect of our involvement with the Games has been the permanent establishment of a West Coast office. dB also now has a new division to handle the design and installation of commercial sound systems. And we have proven that the company has an ability to service world-class special events. There has already been discussion concerning the provision of consulting services for the 1988 Olympic Games to be held in Korea."

Logical Overview

dB Sound maintained 22 salaried technicians for the Olympic Games sound reinforcement project, a figure that included seven "key" people who acted as crew chiefs and supervisors. An informal local labor pool was relied on to some extent, but a majority of the work force were prior dB employees.

"We had a standby system operator on line for every single event," notes Ash. "Paul Brin took charge of 'Control Central' . . . he made telephone calls to check on prompt arrival of the crews at the sites. There were so many variables to keep track of, such as freeway traffic and schedule changes, that we made sure to have a backup crew ready at every venue."

Crew accommodations were varied: motel rooms were rented in San Diego County for the Fairbanks Ranch installation; crew members for the system at Lake Casitas relied on a rented motorhome parked in a campground space; while the majority of the events were serviceable from the Los Angeles area.

Due to the staggered scheduling of events at various sites, system installations did not all occur simultaneously. "We were able to take down one system, sort everything out, and then use the same crew and gear to meet another deadline in many instances," Ash recalls. "We relied on two trucks, three station wagons, and a van to cart the gear around. Only once did we have to schedule a rented Ryder truck."

Communications became the most important part of the operation. "We

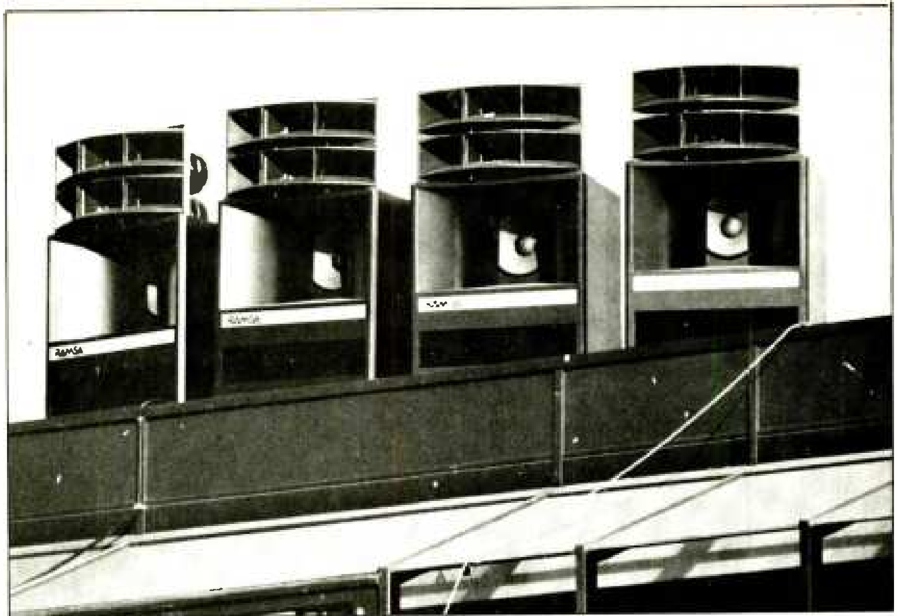

Figure 10: Roof-mounted RAMSA loudspeaker clusters at Pepperdine University's Raleigh Runnels Memorial Pool, where 5,000 spectators attended Olympics water polo events. The sound system was operated by the event commentators.

used Motorola pagers with alpha-numeric displays," the engineer advises. "Messages were passed back and forth between our key people in that fashion. Every person involved in the project called into the central office every day, and the people who had to get up at 5:00 a.m. sometimes were given wake-up calls from the office."

Lessons for the Future

"One thing that really helps a complex project like this is an accurate set of drawings showing the running of various system lines," Ash advises. "At one event site, you would have telephone people. There would be AC

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SOUND FOR THE OLYMPICS

power distribution runs . . . there were network television people, temporary lighting systems, and everything else you can think of. Where trenching was involved for the laying of speaker cable, there were always these other groups of people and their needs to consider. A master line plan, perhaps with transparency overlays for the various systems, is very helpful on a project of this magnitude."

Ash also counsels that negotiating and mediating skills are important ones to have for such a project. "The politics of dealing with so many different production entities could be the topic of a whole book in itself," he laughs! "Having your company's needs met, and still respecting the requirements of others, is all part of the whole project's success. As it was, we were fortunate to have gear to work with that did not break down, and people to offer as operating crews who are used to working in stressful situations. It turned out very well."

dB Sound's Bruce Gordon concurs: "If we had to do it all over again, more advance planning details would have helped. But when those details don't even exist until we actually sit down and figure them out, it just comes down to relying on your instincts and past experiences.

"Security, of course, was a prime consideration. Even though we were taking expensive hardware in and out of 23 separate sporting event sites, three athletes' housing villages, and a number of training sites, our only equipment loss turned out to be four microphones and four casette decks. We feel that the sound reinforcement aspects of the Olympic Games was a complete success." ■■■

BROADCAST AUDIO DISTRIBUTION FOR THE SUMMER OLYMPICS

ABC's Domestic and International Feeds with Digital Multiplex Distribution Techniques and Automated Digital Delay Lines to Ensure Correct Audio/Video Synchronization

by Douglas Howland, consulting editor

In late 1979, ABC signed an agreement with the Los Angeles Olympic Organization Committee and the International Olympics Committee for the exclusive television rights to the Summer '84 Los Angeles Games. In addition to obtaining the domestic TV rights to broadcast Games coverage in the U.S., ABC accepted the responsibility to become the Host Coordinating Broadcaster for the event, and to supply any broadcaster with International television coverage of the Summer Olympics. The basic feed included comprehensive coverage of events at all the performance venues, live or in summary form, totalling more than 1,300 hours of programming. ABC contracted to supply all circuits, facilities, equipment and personnel needed to produce this basic feed from each venue, and relay it to the various World Broadcasters.

A plan was developed in 1980 which, in its basic lines, included the decision to provide an International Broadcast Center (IBC) in Hollywood for World Broadcaster operation, and separate from ABC's domestic unilateral operation based at ABC's Television Center. At each venue, ABC produced at least one World Feed for non-biased International TV coverage, as shown in the accompanying block diagram. Audio and video from the World Feed was transmitted to the IBC, where each individual world broadcaster could create a complete program either live, edited, or pre-recorded, for distribution to its home country.

Broadcast circuits also carried the World Feed from the IBC to the Unilateral Broadcast Center (UBC) for use by ABC. In addition to and separate from the World Feed mobile production truck(s), at most venues ABC provided one or more additional truck for domestic broadcast production. The domestic feed could sample elements of programming from the World Feed truck(s), along with the origination of additional audio and video material. This feed, which comprised two channels of audio — channel #1 as mixed audio and channel #2 as "natural sound" — was routed directly to the UBC, where producers could choose between the two feeds from all sources and create a program for the American audience.

At smaller venues, ABC relied on a local announcer to comment on the event as seen by World Feed cameras. Announcer audio routed directly to the UBC, while the World Feed video looped through the IBC on its way to the UBC. A second video line taken direct from the venue to the UBC allowed an ABC commentator at the venue to go on-tube for the American audience. If the event became more active, however, ABC could have dispatched an additional crew and equipment to feed increased domestic coverage down this second video circuit.

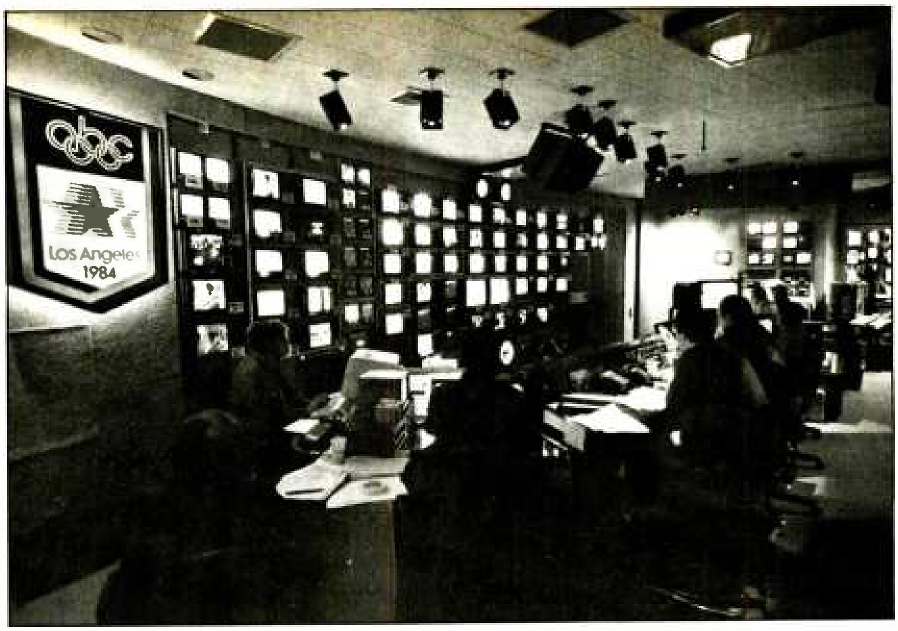
ABC designed, supplied, installed and maintained the IBC technical facilities requested by the world broadcasters. Each one of these facilities was tailored to satisfy individual broadcaster's requirements, although outside broadcasters could also provide part or all of the equipment and services to themselves.

The major technical segments of the IBC complex were divided into four areas: Telco Program operations; Distribution-Synchronization; Transmission Control; and the individual broadcasters' facilities. All program circuits entering or leaving the IBC passed through the telephone company terminating and interfacing equipment, a system that made extensive use of digital fiber-optic, microwave, and satellite circuits, the type of circuit for each event being determined by the venue location and its distance to the Broadcast Center. Next, the incoming audio and video circuits entered the Distribution Center where signals were terminated, routed, synchronized to a master signal reference, monitored and amplified for distribution to each broadcast studio and edit suite. Transmission Control provided the necessary interface between the IBC broadcasters' facilities and the outside world.

To provide foreign-language coverage to the varied audience around the world, commentator positions were provided at each venue to all those requesting such service: over 400 positions were located throughout the many venues. A commentator control unit was custom designed and installed at each position to allow two announcers to function either independently or simultaneously, two dual headsets with noise-cancelling microphones being provided.

Commentators could blend with their audio feeds the natural sound of the event as background. Program feeds were taken directly to the IBC for subsequent routing and distribution. A separate feed of natural sound was also made available to producers at the IBC, where it could be added to the voice reports for added crowd excitement, for example, or subtracted (via phase reversal and cancellation) for increased voice clarity. In addition, talkback to the studio was provided for each commentator, along with a status indicator for "standby" and "ready" condition.

The commentator control unit enabled the various announcers to selectively mix the following monitor sources into the right earphone: Guide 1 or 2; Public address; International Sound (natural event sound); Studio talkback; and Local cue. In the left earphone, the commentator will monitor foldback, local cue, and technical talkback.

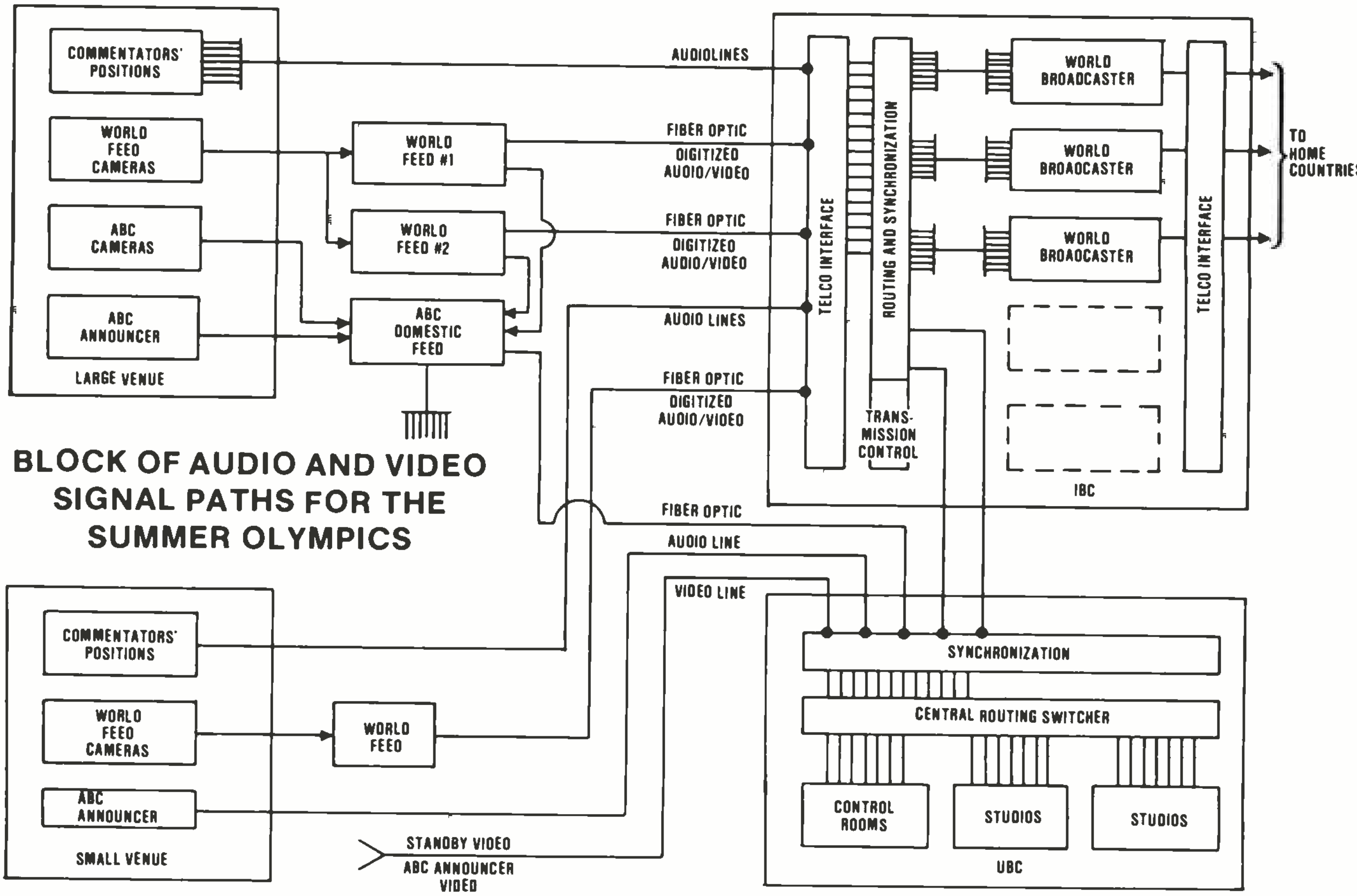


Main Control Room utilized by ABC Television for live and pre-recorded coverage of 1984 Summer Olympic Games.

Stereo Digital Audio Distribution System

In 1979, when ABC began construction of a new technical facility at its television center in Hollywood, the network was also involved in negotiations for Olympic broadcast rights. As a result the decision was made to build a plant capable of fulfilling the complex needs of such a project. The first concern for audio was that the entire operation be stereo-capable for whatever method of distribution was to be selected. Also, a system was needed to route other audio material with the stereo audio feeds, such as second language programming, timecode signals or program cues.

Grass Valley Group developed a new custom-designed unit for ABC, the Model 3280 Digital Multiplexer, which enables four audio channels to be routed down a single coaxial cable; each audio source, VTR, production studio, etc., is equipped with an encoding multiplexer, while each destination is fitted with a demultiplexer. The Model 3280 uses 12-bit digitization,



BLOCK OF AUDIO AND VIDEO SIGNAL PATHS FOR THE SUMMER OLYMPICS

BROADCAST AUDIO DISTRIBUTION FOR SUMMER OLYMPICS

with an additional four bits available for compansion. The result is a device that functions and performs like a 19- or 20-bit system, providing a reported dynamic range in excess of 120 dB. Signals passing around ABC's broadcast center are routed through the second bus of a 192-by-128 video routing switcher that can operate as either audio-follow-video, or in the "break-away" mode to allow independent control. Currently, the four audio channels are assigned by ABC to carry two program channels for stereo, a timecode feed, and the fourth as a "spare," or undefined purpose.

According to ABC's Dave Elliot, general manager of UBC operations/engineering, the benefits of such a digital system are many. First, a single coaxial cable does the job of four individual shielded pairs, with a resultant saving in cable and labor costs throughout a system this large. Second, crosstalk and noise are virtually eliminated. Also, the system is much more adaptive to change: A circuit that once carried four channels of audio could easily be assigned in the future to handle a video signal.

Since a great number of commentator units and program circuits were in use at numerous venues, a means of quickly identifying program circuits and announce positions was considered necessary. An

inexpensive cassette player and endless-loop cassette enabled each individual program line to be identified by its own "personal" voice. At the end of an event, a switch was toggled from the "operate" position to "ID."

A major concern for Olympics' coverage was to ensure that the video and audio signals were synchronized in time. "While people will accept psychologically seeing the puff of smoke from the starting pistol, and then hearing the shot a little bit later, the mind gets really upset when they hear the shot before they see the puff," confides Chris Cookson, director of ABC's Olympic Centers.

Video delays can be caused by insertion of frame synchronizers and time base correctors into the signal path between a remote site and the broadcast center. (Such devices are necessary to ensure that all the frames of a remote video signal start at the same time as the house reference, to prevent picture shifts and rolls when switching between various in- and out-of-house signals. Also, differences in audio and video path length can cause timing offsets between the two signals.

Obviously, an audio delay device is needed to maintain synchronism between sound and video. While manually adjustable, fixed-delay devices are available, ABC felt the need for greater sophistication and control over the synchronization problems. As a result, Tektronix developed a custom designed Automatic Variable Digital Delay unit for the network to work in conjunction with ABC's frame synchronizers, a total of 57 such units being used during the Summer Olympics. An RS-422 interface connects the video (frame sync) and audio delay units together to ensure proper tracking.

The control signal consists of timing information obtained from the frame synchronizer, and represents the difference between a clock timed to the incoming video and that at the output compare to house sync. According to Chris Cookson, the frame sync unit controlling the Tektronix AVDD "reads in a clock timed to incoming picture, waits, and then reads it out in time with house sync. The difference between their two synchronizing sources tells [the AVDD] what delay to put it."

The delay unit is said to be inaudible in operation during normal programming, even when the change is from minumum to maximum delay and back again; on a steady-state tone, however, it is possible to hear a slight warble. The unit works by altering only very small segments of the audio — typically between five and 10 microsecond pieces — and waiting for a small time interval before manipulating the next piece. For audio that needs to be delayed by, for example, four video fields (66 milliseconds) the delay would build up over a period of five seconds. ABC points out that it would be very rare to encounter this large of a timing error; normally, the unit is expected to pass only through a few lines of video, and at times a field's worth.

Undelayed audio was also made available to producers for the primary purpose of an "in-time" return feed to the live-event announcer headset, and was carried between studios on the fourth channel (spare) of the multiplexer.

All Olympics audio signals were routed to a premix/selection studio for distribution to Control A and Control B, two control rooms that served as the final audio feed point to the network lines — Control A was used during prime-time hours while Control B served all other times. These two, virtually identical studios also served as backup to each other in the event of equipment failure. ☐☐☐

RECENT ADVANCES IN SERVO-CONTROLLED CONSOLE AUTOMATION SYSTEMS

GML MOVING FADER AUTOMATION SYSTEM

by Denis Degher

With computers finding their way more and more into our daily lives, it is no wonder that they are becoming "must have" systems in the repertoire of contemporary recording technology. With the ever increasing amount of tracks available on today's sessions, it is becoming physically impossible to mix modern music with the two appendages known as hands. In film and video post-production houses, the ability of computerized audio consoles and automated video switchers to speak a common language may prove to have even more beneficial results in the future. Synchronizing ADR (automatic dialog replacement), sound effects, and audio in Music Videos should become more than just a trial and error event.

Automated consoles are also helpful in ensuring intentional placement of a part in the mix without resorting to the use of dynamics-robbing devices such as limiters. From an engineer's standpoint, automation can be very beneficial in trying to catch that particular guitar part that you enjoyed during the tracking, but now find somewhat buried in synthesizer overdubs. Virtually any type of nuance and texture can be created with repeatability, providing us the opportunity to mix in a static situation.

Although automated consoles have been around for nearly a decade now, in my experience, many engineers are still skeptical of their applicability, and some even consider automation to be unnatural and unmusical. Others state that their reason for disaffection with console automation stems from its association with the VCA (voltage-controlled amplifier) as the gain-change element, and its subsequent signal degradation. A valid complaint with most VCA/tape-based automation is the accumulating time lag of all fader moves and channel mutes with each ping-ponging stage between pairs of data tracks. Many engineers feel they can accomplish the same end result as tape-based automation by mixing sections and putting their mixes together via razor-blade edits.

Automation systems of the past have had various drawbacks, predominately reliability, and were, for the most part, tolerated. But engineers that have been successful with automation will agree that its ability to control the dynamics of the mix are undeniable, creating a much more relaxed atmosphere during mixing, and that programmable muting enhances musical arrangements as well as vocal switches. Major fader moves, as well as textures and nuances, can all be achieved with repeatability, freeing

. . . continued on page 86 —

NEVE NECAM 96 SERVO-FADER AUTOMATION SYSTEM

by Morgan Martin
regional manager, Rupert Neve, Inc.

In 1976 Neve introduced the Neve Ergonomic Computer Assisted Mixing system, or NECAM, which was intended to represent the ultimate in a "user-friendly" console automation system. Thus the main control box employed dedicated keys with common names like PLAY, GO, MIX, FROM, etc. In addition, NECAM was designed without VCA's to avoid their attendant problems of noise, drift, distortion, and grouping inaccuracy. As by now is well known, instead of a VCA, the NECAM fader uses a standard conductive-plastic track, controlled by a servo-driven motor. The reason for this approach was simple. In a VCA-based system there are really two faders in each channel: the "real" fader that the operator uses during a mix session; and an "imaginary" fader that the computer uses to play it back. The problem comes when the operator wants to make an update, since with most VCA-based systems he or she must make the position of the real fader match that of the imaginary one, by using dedicated LEDs or meters, bars on a video display unit, auto-null techniques, or whatever.

On the other hand, NECAM's touch sensitive moving fader makes an update trivial; since NECAM moves the real fader that the operator uses, they are *always* nulled before an update, and there is no imaginary fader to be nulled, nor read, write, or update switches to worry with.

NECAM also provided control of the tape or film transport with manual control or automated locate with up to 999 memorized locate points. The control of the transport was via the main control box using keys labelled PLAY, GO, FROM, TO, etc. It was also the first system to store mix data on floppy disk, instead of the multitrack. Storing data on tape severely limits the total number of mixes that may be kept (usually two, sometimes eight) and, because of record/play offsets during each update, will suffer from drifting mutes after four or five passes.

In about 1981, Neve introduced NECAM II, which added a number of features required for post-production work, but retained all of the features of the original system; the latest generation system, NECAM 96, is intended to provide the operator with even more facilities.

System Capabilities and Operator Interface

The new system will control up to 96 moving faders, twice as many as NECAM II. Also introduced with NECAM 96 is

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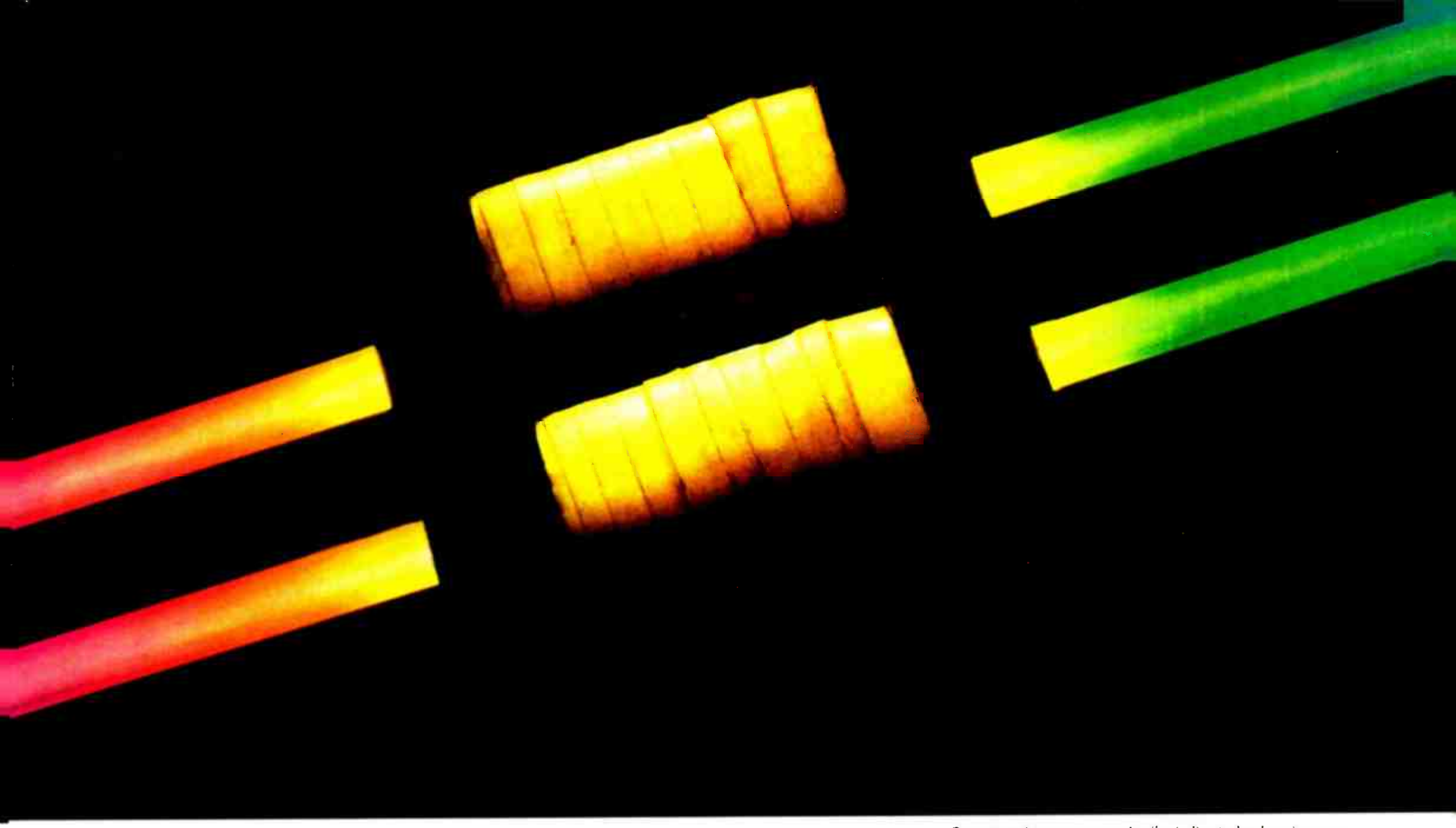

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NEVE NECAM 96 SERVO-FADER AUTOMATION

a new flat-belt fader that offers a feel rivalling that of a manual fader, and which moves from top to bottom of its travel in just over 0.5 seconds — about three times faster than faders of just a few years ago.

NECAM 96 is now equipped with a new control box and a new display. All the keys from previous NECAMs are included, along with several more, each with a key legend that tells the operator exactly what it does. (For instance, NEW LABEL, not unsurprisingly, creates a New Label.) A separate color video monitor is used to display menus and operator instructions. For example, the Main Display provides the data disk name (perhaps a song title or the project name); mix information; tape machine status, etc. The LABEL area shows the number and timecode location of the last label passed and the next one coming up. A standard QWERTY keyboard enables mixes, labels, stores and function switches to be named, and extensive notes made about the session. In fact, if the LABEL key on the control box is pressed, the display shows the entire label list, highlighting the one just past, while pressing MUTE displays all of the mutes that have been set up. Each entry shows the channel number, whether the channel was muted or turned on, and the timecode location at which it happened. In addition, a mute can be trimmed without having to do it again on the fly; from the MUTE list display the time key can be used to trim an entry to within a quarter-frame (25 fps) or a third-frame (30 fps). (Note that whereas other systems sample mute activity on a subframe basis, NECAM 96 actually allows these mutes to be trimmed on a subframe basis). A press of the MIX key displays a list of all the mixes stored on the data disk, along with FROM and TO labels and information about each mix.

The system also provides up to 128 event switches, each of which controls a relay that can be used to operate external equipment, such as cart machine starts, roll in machines, ABC transfer keys, bypass keys, etc. The EVENT lists show all of these switch operations in timecode order, and, as with mutes, allows subframe trimming and off-line entry. This capability is useful in post-production sessions, where a list is usually prepared in advance of the sweetening date indicating the timecode at which various effects are to roll in. The times can be entered via the keyboard, and then trimmed as needed during the session. (They can be done on the fly, of course.)

In addition to using SMPTE/EBU timecode for system reference, NECAM 96 accepts machine tach pulses for use during locates, so it is not necessary to have tape in contact with the heads during wind mode. The system also accepts foot/frame counts instead of timecode, all times then being displayed in feet and frames for film re-recording sessions, etc.

Snapshot Mixes

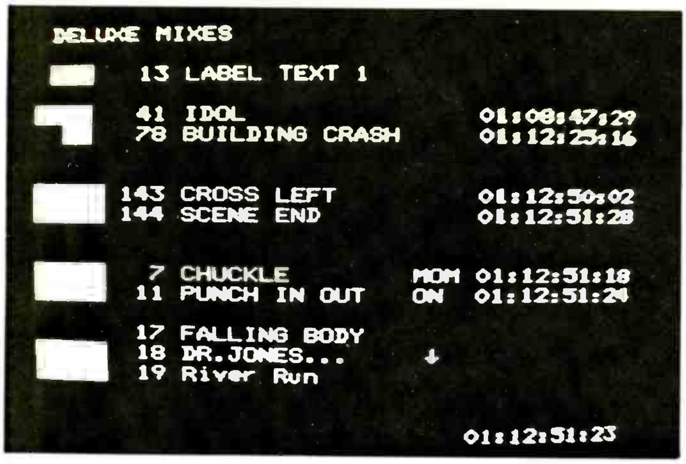
A STORE is a snapshot of the positions of all faders, mutes and event switches, and can be called back to the desk at the press of a button. In contrast to NECAM II, which offered one store, NECAM 96 offers hundreds of stores, any of which may be set to the desk by one of three ways. Press SET (as in NECAM II) and the STORE specified will be set to the desk instantly. Pressing AUTO XFADE will cause the faders to crossfade over a period of time — designated by the operator — from their present positions to the STORE positions. (This can be useful in merging two mixes where faders most likely are not matched at the Merge Point.) Alternatively, the MANUAL XFADE enables a crossfade to be made to a STORE at any selected rate; for example going half-way over three bars, and the rest of the way over two beats, via the Manual Crossfader.

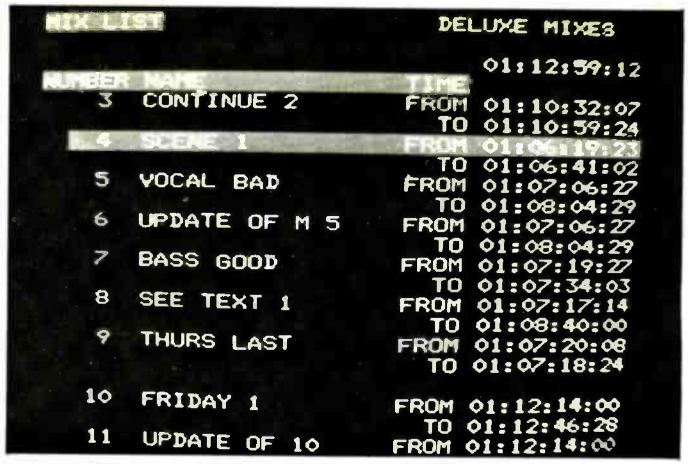
A STORE list shows all the STORES that have been created, and allows them to be named. The names can act as cues during a mix. For instance, in post-production if a STORE is to be manually crossfaded to when the spiders start to crawl up a damsel's arm, this STORE might be named "SPIDERS" (or "UCK!") as a cue for the operator. Similar cues could, of course, be used for the names of labels.

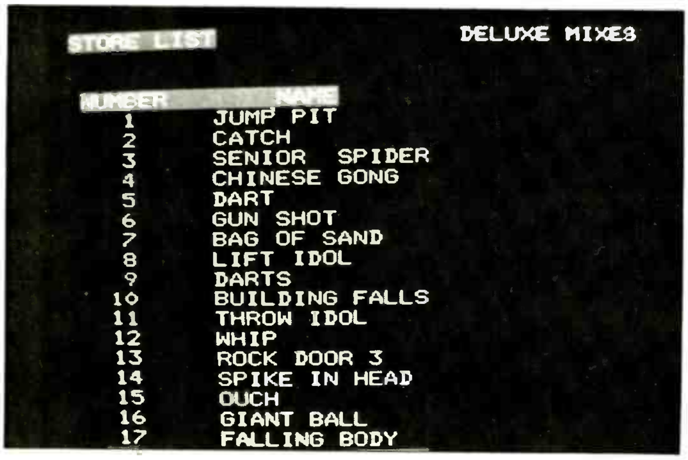
Grouping and Data Storage

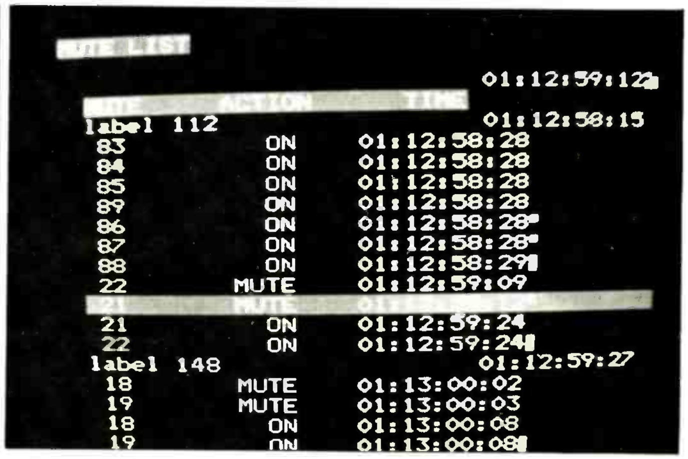
NECAM 96 offers the same free-grouping system as NECAM II, where any fader can be designated the group master rather than having to use control submasters located at the other end of the desk. (But for instances where they are handy, a central submaster fader option will be available shortly.) To set up a group in NECAM 96 it is not necessary to take all the faders out of manual; the operator

NECAM 96 fitted to a Neve Model 8128 console, showing new control unit with additional features, and full-color video display.









Typical NECAM 96 Display Screens *(clockwise from top left)*: **The Main Display, showing name and numbers of mix called from disk, with label and event locations, plus current timecode; The Mix List of mixes kept on disk; The Store List that shows name and number "snapshot" stores; and The Mute List for current mix, showing which channels turn on at what timecode location, to the nearest one-third frame.**

simply presses GROUP, and touches the required faders.

The moves of a fader set to group are individually written, so that if the group is disbanded the moves of each fader are still in memory. In other systems, group moves are kept as one fader, causing problems if a fader is to be used to put down the basic moves as a group, and then the operator goes back to work on individual faders of the group.

With NECAM II, a Mix had to be kept on a client's 8-inch floppy disk data, in order to play it back to update it. Thus the sequence would be: make a pass, keep it, play it back for updates, keep that (Mix #2), play back Mix #2 for updates, keep that, and so on. Each of these "keeps" involved using up space on the data disk. In addition, the second and third updates were probably not much different from the original, but each had to be kept so they could be updated.

NECAM 96, on the other hand, handles data storage a little bit differently. The first mix pass is written onto the scratch disk; to update it, the mix is played back from the scratch disk, run through the console for updating, and then sent back to the scratch to be recorded in another area. At this point on scratch will be recorded the update and the previous mix upon which it was based. Normally, the update is better, and the operator will wish to perform an update of it. Simply pressing LOCATE PLAY will cause the tape to locate to the top of the mix, and play back the update for more updates. At the same time, the "previous" mix will be dropped off. At the end of the pass the first update and the second update will be held on disk. Normally, the first update will be dropped off and the second update will form the basis of the next one.

As can be seen, the data recorded to the scratch disk will always comprise the mix that was just made (update mix), and the one upon it was based (previous mix). Any number of progressive updates can be made, dropping off the previous mix in turn each time, all without the necessity of pressing KEEP. Of course, after several updates the operator might want to KEEP your latest update as a milestone. Simply pressing KEEP causes the data to be transferred to the data disk. Since it is not necessary to have to keep each pass, and since the NECAM 96 data disk holds twice as much mix information as with NECAM II, it is expected that disk changes will be required far less frequently than with previous NECAM systems.

Several modes may be used for the faders. The "Normal" mode has been described above: Just touch the fader and it is instantly in update mode; release the fader and it will play back the mix. Manual mode causes previous moves to be ignored, and new moves written. Relative mode causes the fader to play back the moves with an overall change in level, up or down, as set by the fader. Alternatively, Suspend mode enables the Previous mix or its Update to be played back; in this case the faders may be moved experimentally, as it were, without updating.

ROLLBACK is used whenever the operator needs to go back in the mix to catch a move that was missed, for example, or to go over a section several times to make complex build-ups. Pressing the ROLLBACK key causes the tape or film to wind back any preset amount of time (say, 10 seconds) plus whatever pre-roll may have been specified. NECAM 96 quickly sets the mutes and faders from memory to where they were at that point in time.

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NEVE NECAM 96 SERVO-FADER AUTOMATION

PLAY is automatically resumed and the recent updates that had been made are played back. Just by touching and moving any fader, the operator can add any moves and change any that have been made in the other pass.

Consider a situation during a TV sweetening date for example, while recording to three- or four-stripe, or a composite track. If the producer subsequently needs a laugh track to be a bit louder, the operator simply hits ROLLBACK. The tape rewinds 10 seconds plus pre-roll; the machines go into play mode and synchronize; and the fader levels and mutes reset for where they were at that time. Since the faders are precisely set and begin to replay the previous Mix, the operator can go back into record immediately without any A/B tape-in/out comparisons.

Close up detail of NECAM 96 control panel and keyboard.

User-Defined Key Sequences

Another feature of NECAM 96 is the SMART KEY, which causes several combinations of keys to be pressed simultaneously. For example, pressing SMART KEY #10 is identical to pressing the following keys in turn: LOCATE; "-"; "1"; "0", "0"; "0"; PLAY. (In fact, this SMART KEYS #10 is labelled ROLLBACK on the key cap.) The nine other SMART KEYS can be set up virtually any way the operator wants. For example, you might want a key that actually rolls forward to skip over standard commercial breaks in a program. Suppose that the commercial breaks are three minutes long; SMART KEY #9 could be set up to be: LOCATE; "+"; "3"; "0"; "0"; "0"; "0"; PLAY, which would cause the tape to locate forward by three minutes and resume PLAY and mixing. Or maybe the operator wants to set up a SMART KEY that says "Forget about the updates I just did — Playback the previous mix to be updated again." This SMART KEY would be the same as: PREVIOUS; LOCATE; FROM; PLAY, which represents a savings of three keystrokes. Of course, if necessary, a SMART KEY cap can be engraved to match whatever it has been set up to do; the last example might have a key cap labelled "DUMP UPDATE."

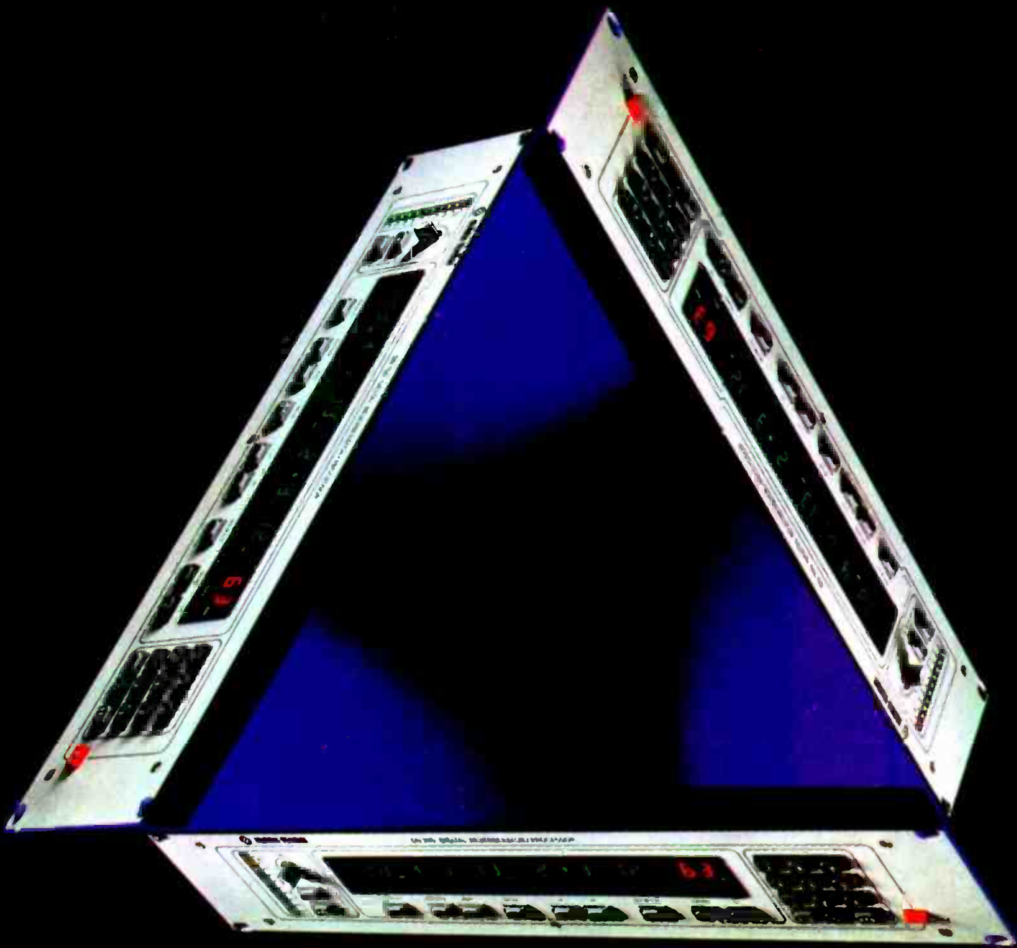
Merging Mixes

Another feature of NECAM 96 is its ability to wind or rewind manually to any point of the tape, and resume mixing, merely by pressing PLAY. The system will set the faders and mutes for that timecode location, and play back any mix information for updating. Whereas the designers believe that the combination of this facility and the ROLLBACK key will result in the less frequent need for MERGE, NECAM 96 also offers considerable changes in this area. First off, NECAM 96 is given all the relevant instructions before it actually does the MERGE. At the designated points, the Merge Function will enter the mix with faders set to the corresponding positions for that point in the mix. Thus, if Mix A is being joined to Mix B at Label #1, the system will determine where the faders are at Label #1 of Mix B and use those settings at the Merge point, Label #1. Should any faders be at different levels on either side of the merge point, the system will crossfade as needed over any preset length of time you like. Or, by instructing "crossfade time equals 0," the faders can be made to jump to their new positions at the Merge point.

. . . continued overleaf —

A small universe

325
164



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9653 Côte de Liesse/Dorval, Quebec H9P 1A3, Canada. Telephone: (514) 636 9971

For additional information circle #49

Regarding machine control, should some other device (for example, a CMX video editor) be required to control the transport, NECAM 96 is simply set to run in SLAVE. The system will respond to incoming timecode and mix as normal; it just won't try to control the transport. Or, in the other direction, Neve will optionally provide an interface to popular synchronizers so that NECAM 96 will "talk" to them as if they were a transport, providing data on the desired locate point and letting the synchronizer perform location of the various machines. Thus chase operation would not be required.

For those interested in hardware, the computer Automation LSI computer with core memory — as used in NECAM I and II systems — is replaced by a CompuPro microprocessor with CompuPro RAM memory cards; these items are available over the counter to any buyer. The drives used may be those of existing systems or, for new systems, dual 8-inch Shugarts in a Neve-built frame. In general, the existing peripherals can be used with little or no modification; existing faders may be used, if desired. In addition, a conversion copier is supplied to convert the single-density NECAM I and II disks to the double-density NECAM 96 disks. ■■■

GML MOVING FADER AUTOMATION SYSTEM

— continued from page 78 . . .

one's hands for echo, panning, and other effects moves.

While both servo-fader systems and VCA-based automation systems have their drawbacks, they also offer significant advantages as well. A major advantage of the Neve NECAM system is its use of motorized faders that offer visual confirmation of level changes controlled by the automation computer, and the elimination of VCAs. An advantage of VCA-based automation systems is that the grouping and group muting can function directly without accessing tape- or floppy-disk-based information tracks.

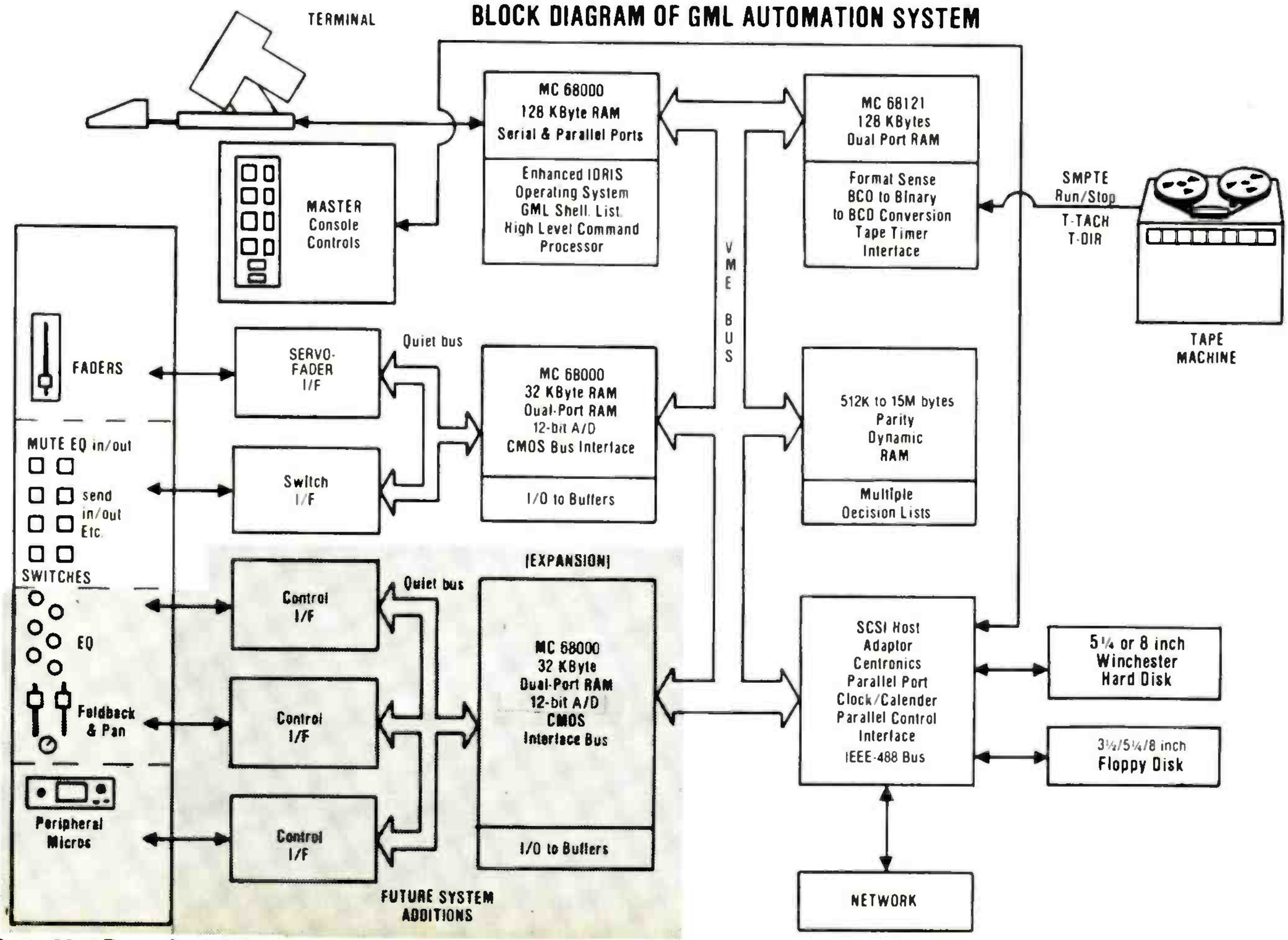
Recently, a second-generation moving-fader automation (MFA) was introduced by George Massenburg Labs, and installed, among other places, at Conway Recorders, Hollywood. Although the system had been in operation for several years at The Complex in West Los Angeles, the Conway installation marked GML's first foray into marketing a production model of the system.

Engineer/Computer Interface

According to its designers, the new GML system attempts to address the automation issue from the engineer's perspective, and extends the operational advantages of both MFA and VCA automation using fast, reliable, relatively inexpensive computer technology. The system utilizes multiple off-the shelf MC68000 processors, and is commonly configured with 512K of random access memory (RAM) for storage of large real-time decision lists. Operating systems, user-command interfaces and utilities are configured to run in an additional 160K of RAM. Automation data is written in 68000 machine code for speed, with user-interface software written in C language for ease of custom revisions and updates.

While the automation storage system may vary with applications, in essence the mix data, programs, operating system, and utilities are stored on a Winchester hard disk. Several storage formats are available from GML, ranging from a 10-megabyte, 5¼-inch format, to 80 megabytes on an 8-inch format. The Conway automation system, for instance, uses a 20-megabyte hard disk; according to

BLOCK DIAGRAM OF GML AUTOMATION SYSTEM



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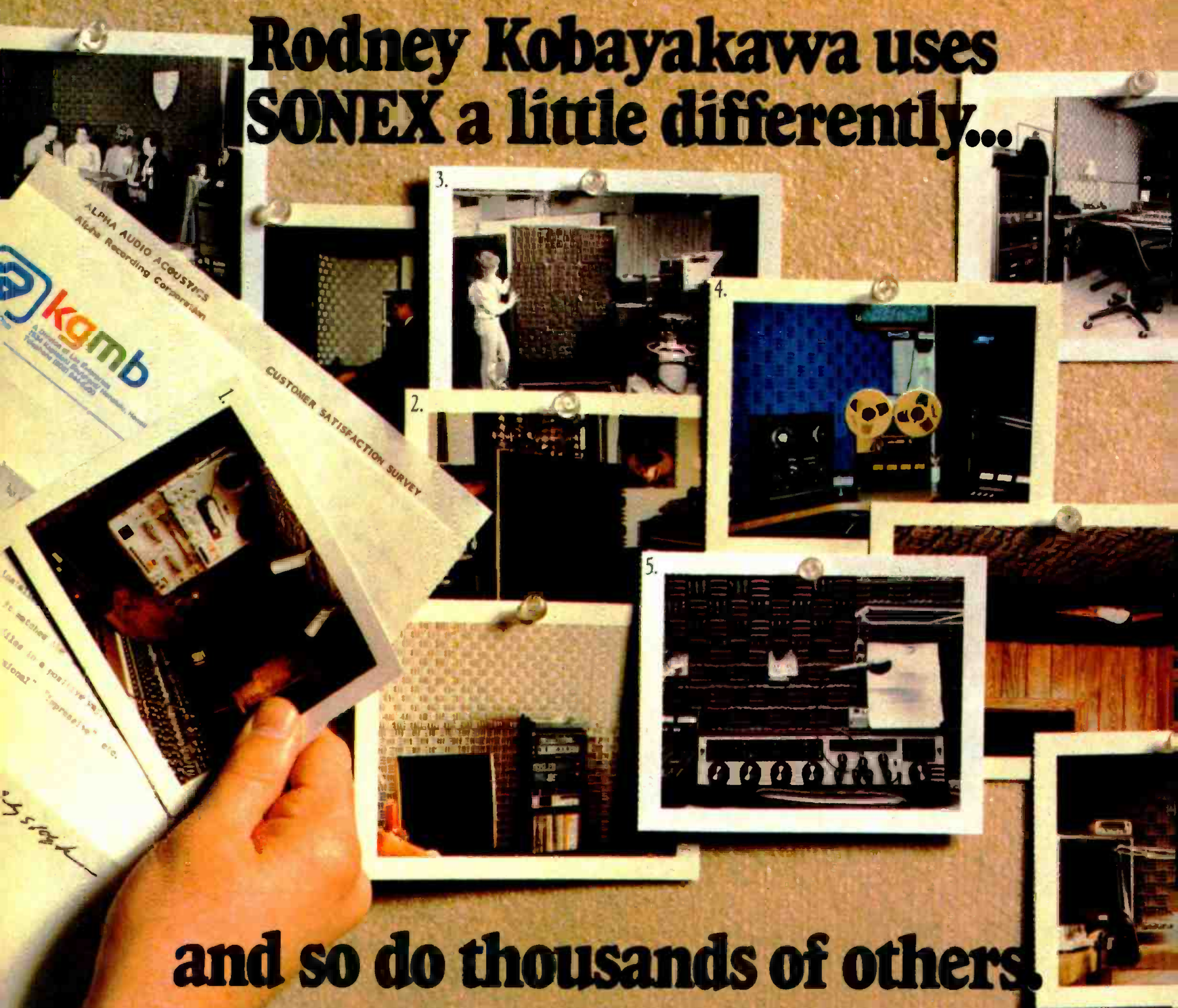
Rodney Kobayakawa uses SONEX a little differently...



and so do thousands of others.

Just Listen — 1. You might even say that Rodney Kobayakawa's use is off-the-wall. He lines the ceiling of KGMB-TV's remote video truck with SONEX. "Controls the high ambient noise…Close speaker mounting caused problems with sound orientation…and they look so nice! White audiotiles match the chrome look of our equipment." He's the Audio Engineer at KGMB, Honolulu.

2. "Musicians are always impressed by your SONEX products." Dennis Scott of Chelsea Entertainment Organization, who won a Grammy in 1981 for **Sesame Country**, an album featuring Crystal Gayle, Loretta Lynn, Glen Campbell and the Muppets.

3. "Takes the 'ping' out of hard walls." Don Bachmeier, KFYR-TV, Bismarck, N.D. uses SONEX on wheeled, portable panels to kill voices on adjoining news sets.

4. "Delighted with SONEX's effectiveness…pleasing aesthetics…audio professionals notice reduced standing waves…increased soundproofing." Sherrie Thomas, Producer, recording studio for the General Conference of Seventh-Day Adventists, Washington, D.C.

5. "Good!…in radio station's master control and production room." Craig Falkenstine, WJCF, Morgantown, W.V.

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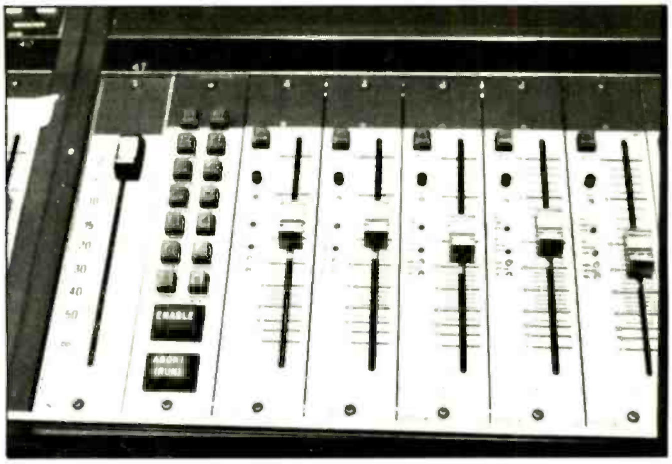

The first production GML Automation System installed to a 48-input Neve 8108 console at Conway Recorders, Hollywood. Shown right is a close up detail of the master select panel and a section of the input fader bank.

studio owner Buddy Brundo, "We have only utilized 40% of the hard disk in four months, so we haven't had to rebuild [purge unwanted files from] the system from hard disk to floppy yet."

A floppy-disk system is available in either a 5¼- or an 8-inch, double-sided, double-density version, and is used to archive mixes. The 8-inch floppy hardware may also be used to read data recorded on NECAM or Solid State Logic automation systems.

One of the most popular features of VCA-controlled consoles had been the ability of the automation to enable subgrouping, group-muting, and soloing. Most can operate without external intervention and without operation of the multitrack. In this respect, the GML system facilitates grouping and group muting without rolling tape, by manipulation of the data stored in on-board RAM.

Like other SMPTE-based automation that use a disk medium for storage — for example, the SSL and NECAM systems — one track on the multitrack or VTR must be utilized for SMPTE timecode. Unlike other disk-based systems, however, the automation is configured to operate as a slave system that follows the SMPTE code (or, in the case of film transports, the tach timer) from the master machine. In this way, control of the entire automation system is enabled via master tape-machine controls, facili-

. . . *continued overleaf* —

A typical display screen menu (left), plus Conway's keyboard control surface and video display unit. The keyboard (shown in greater detail below) enables labels and annotations to be added to the mix data, and provides enhanced system control.

GML MOVING FADER AUTOMATION SYSTEM

tating quick and random updates anywhere on the tape. By comparison, other systems must command the tape machine; an update can begin and end at any new created "label" section, but the song must be assembled section by section for *each* modification, a procedure that can be laborious and time consuming.

The GML system holds the entire contents of the current decision list, which contains the fader dynamics and switch status data, in on-board RAM, and requires no section labels. Updates in the play mode are achieved by winding directly to the update area, at which point a software phase-locked loop is started and the console reset to its proper position at that timecode location on tape. Alternately, fader and switch data (mute, EQ in/out, send in/out, etc., up to total of eight on/off functions) may be written statically as "presets," and precise mix changes instituted at a specific SMPTE address, from a cue point in a song, or a CMX decision list, etc. Few computer keyboard entries are required, the designers state, until a mix is to be stored on the hard disk.

The GML DC servo-controlled faders feature touch-sensitive, chrome-plated fader control knobs. A fader touched with a fingernail or pencil eraser, for example, will not activate the fader-touch electronics; instead, the move can then be completed by touching the fader with a finger, allowing tight cueing. The GML faders have been designed to be extremely light to the touch — creating a total mechanical resistance of less than 80 grams — to provide the feel of a standard, non-motorized fader.

What prompted George Massenburg to design and build his own servo-controlled automation system, we asked? "I needed an integrated system for ARC studios [later, The Complex] that was at least as good as the custom audio electronics we were building at the time," he recalls. "Our system is friendly, 'screen-cuddly,' very fast and powerful, and *not* running anywhere near close to its projected performance limits."

The system is configured to run four times faster than SMPTE timecode, and is able to turn mute and seven other assigned switch functions per channel on and off every quarter frame.

"We planned on the Motorola

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The total freedom of movement promised by wireless mic and instrument systems is a very attractive option for most modern musicians and singers. Unfortunately, the stratospheric prices of the more "professional" systems exceed their ability to eliminate problems like *drift*, *interference* and *dropouts*—so many of the most adventurous performers have gone back to messy cabling.

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The system consists of a full-frequency response microphone (with a choice of the Shure SM58 Dynamic or SM85 Condenser capsule) or "belt-pack" instrument/lavalier transmitter and Samson's PR-50 30 Channel FM Digital Receiver. These elements working together result in a frequency response of 20 Hz to 18 kHz, extremely low harmonic distortion and excellent signal-to-noise specs for the strongest, most uncolored and "natural" sound possible. The system's range is a full 300 ft. under the most adverse conditions and 1500 ft. line of sight.

To offer greater performance, the Samson PR-50 receiver features an exclusive three-parameter "Auto-Mute" (patent pending) function to mute off-station noise when the mic is off; a streamlined and compact 19" rack-mountable housing and balanced line outs and a line level out for recording and mixing. Samson's HT-20 microphone transmitter uses a standard 9V battery and has *no* protruding antenna to get in your way.

And when it comes to the *bottom line*, Samson's Broadcast Series System is priced at $1,295 U.S. (w/Shure SM58 mic cartridge), $2,000 less than our closest performing competitor. In fact, competitive units at twice the price don't offer all of these features. We've created *the* system that solves your biggest problems with wireless. The Samson Broadcast Series "Phase-Reflex" Wireless System is the first *real* step towards experiencing the ultimate freedom of wireless.

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For additional information circle #54

MC68000 family [of microprocessors and support chips] in 1979," Massenburg continues, "long before cheap systems became available. We liked the things that Motorola were saying about software upgradability, some of which turned out to be true. Our system is updatable, allowing relatively graceful expansion to more and more functions like EQ, foldback, and panning through ports that are pretty much already in place [see attached block diagram].

"Our design and manufacturing philosophy was to not merely build an adequate or competitive system, but to build performance and quality. Originally, we had wished only to design the system and write software, but we were obliged to manufacture a good deal of hardware as well because it simply was not, and was not going to be, available."

Automation Case Example

The first production GML automation system is now resident at Conway Recorders, retrofitted to the studio's 48-input Neve Model 8108 console. According to owner Buddy Brundo, "The GML system was designed with the needs of today's engineers in mind — after all George is himself a recording engineer as well as a design engineer, and he built his system so that it is utterly simple to operate.

Main GML computer and disk drives

"It totally frees you from the mechanics of the mix: you can infinitely adjust levels, merge mixes, and create a complete new mix at any SMPTE address. Ergonomically speaking, everything has been taken into consideration; the system has been running flawlessly for [several] months.

"The phone has been ringing off the wall since the system has been installed," Brundo continues, "We have mixed the dance version of Billy Idol's *Flesh For Fantasy* with Keith Forsey producing; mixed the new Motel's album with Richie Zito producing; and mixed former Average White Band member Alan Gories' solo project with Jay Gruska producing — Mick Guzauski engineered all the projects. Engineer Frank Byron Clark has also returned with producer Reggie Andrews to mix the new Dazz Band album."

Since the Conway installation, Ocean Way Recorders, Hollywood, has added a GML automation package to its 48-input API console, and ordered a second system for a 48-input custom Delcon/API board. At the time of writing, orders for Mama Jo's Studio, North Hollywood, Lion's Gate Films, West LA, Record Plant, NYC, and Bill Schnee's Studio are pending.

Having developed its own moving-fader automation system, GML is currently at work with several major console manufacturers, including Soundcraft and AMEK, to integrate automated panning, equalization, routing, and auxiliary sends into affordable recording consoles. Since the GML system has a *general* system architecture and protocol, it can already communicate with future microprocessor-controlled outboard devices; it is only a matter of time before many other redundant tasks become automated, its designers say.

■■■

TECHNICAL OVERVIEW OF GML MOVING FADER AUTOMATION SYSTEM

The system includes:

- CPU — dual Omnibyte MC68000 processors, system control processor (MC68121) and hardware SMPTE reader, SCSI I/F, 512 Kbyte external dynamic RAM, peripheral interface, 20 Mbyte 8-inch Quantum hard disk and 8-inch floppy.
- Servo-motor driven fader #8400D (with P&G Model 1500 D21216 fader).
- Fader Interface and Servo Amplifier #8320.
- Switch Interface #8324.
- Master Control Panel.
- Freedom 200 (or other generic) Video Terminal.

Specifications:

1. Dynamic Accuracy — Full fader travel in 80ms to 1% of desired target, with less than 1% overshoot (critically damped); settling to 0.1% of desired target in 250 milliseconds.
2. Theoretical minimum granularity — (software) 0.09375 dB; (hardware) ± 0.1% total full scale.
3. Total relative accuracy (single fader repeatability) — ± 0.125 dB from 0 to -20 dB; ± 0.250 dB from -20 dB to -40 dB (depends solely on fader element).
4. Total absolute accuracy (fader to fader) — ±1.0 dB from 0 to -20 dB; ± 2.0 dB from -20 dB to -40 dB (depends solely on fader element).
5. System timing performance — Faders are scanned every frame (33.33 milliseconds at 30 fps) and refreshed every quarter-frame tick, (8.33 milliseconds at 30 fps); switches (mutes) are scanned and refreshed every tic (8.33 milliseconds at 30 fps ± 2 milliseconds).
6. System architecture and input/output topology allow for gradual expansion to 128 channels of 16 words (X16 bits) of data per channel, real time (i.e., per tic), and an additional 128 channels of 16 words per channel static, or set up data — a total of 128 channels of 32 words per channel. Fader data are stored on the decision list in vector form as follows: target dB value, 0.09375 dB steps from 0 to -88.5 dB or 10 bits; vector interval, 1 to 63 frames or six bits — a total of 16 bits. Switch data are stored as eight bits of switch status, and eight bits of command, to allow merging of individual switches. Other data such as EQ and foldback are to be stored as lower priority, with data packed into 16-bit words.
7. System timing reference — SMPTE input is phase-locked to an internally generated system clock running at four times SMPTE rate (120 Hz for 30-frame non-drop, 96 Hz for 24-frame code).
8. System refers to tape timer and tape direction signals to keep track of tape position when not playing.
9. The fader electronics generate a signal upon touch activation. The touch electronics are stable, calibration-free, and react under any or all of the following conditions — 500pf nominal body capacitance; a nominal body-field of 50-60 Hz AC; 5 kohms DC resistance to ground. □□□

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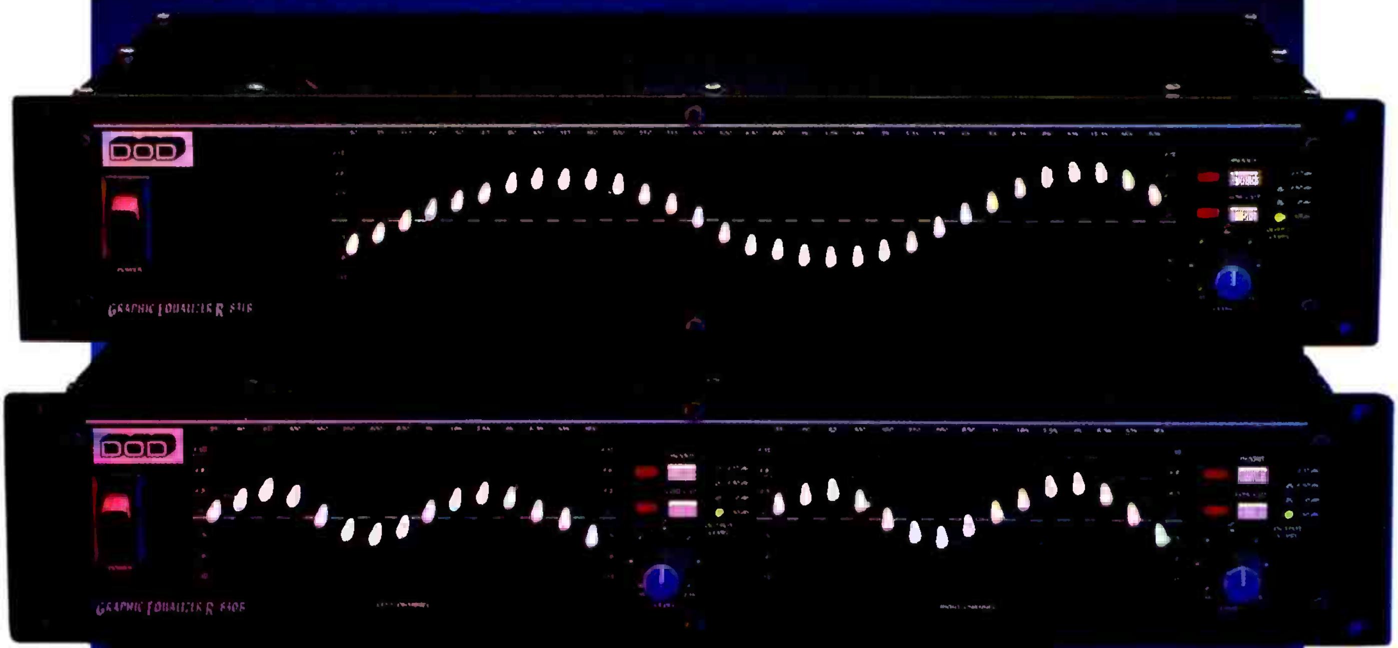

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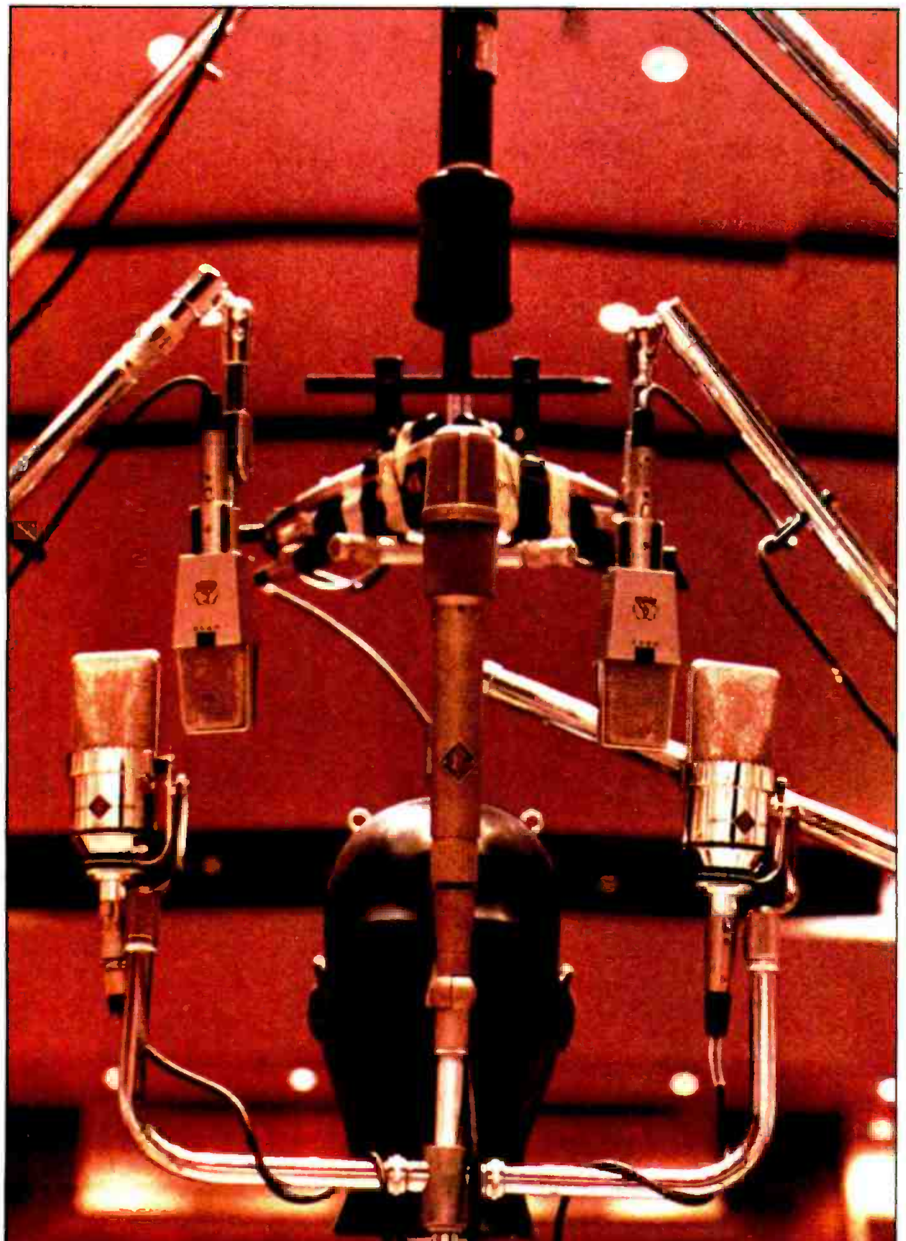
For additional information circle #55

The first in this series of experimental recording sessions, conducted at The University of Iowa School of Music and described in the April 1984 issue of *R-e/p*, provided a comparison of 12 types of microphones recorded simultaneously as stereo pairs. Additional information about the test conditions, as well as a dialog concerning the subjective element in undertakings such as this, can be found in an interchange between John St. John and the author in the "Letters" section of the August 1984 issue of *R-e/p*. Readers are urged to consult those two issues for details relating to the musical performances, the concert hall, and the equipment and procedures of the earlier evaluation.

PERFORMANCE ASSESSMENTS OF STUDIO MICROPHONES

Additional Stereo Evaluations of 15 Models at The University of Iowa School of Music

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



Photography by James. J. March, Linda Bourassa and Steven Sergeant

The initial article closed with an invitation to manufacturers and importers to make their products available for further comparisons. I am very gratified by the generously cooperative (and at times even good-naturedly insistent) responses that I received from the various individuals and firms whose products are included in this second round of evaluations; without their cooperation the present report would not have been possible.

Recording Sessions

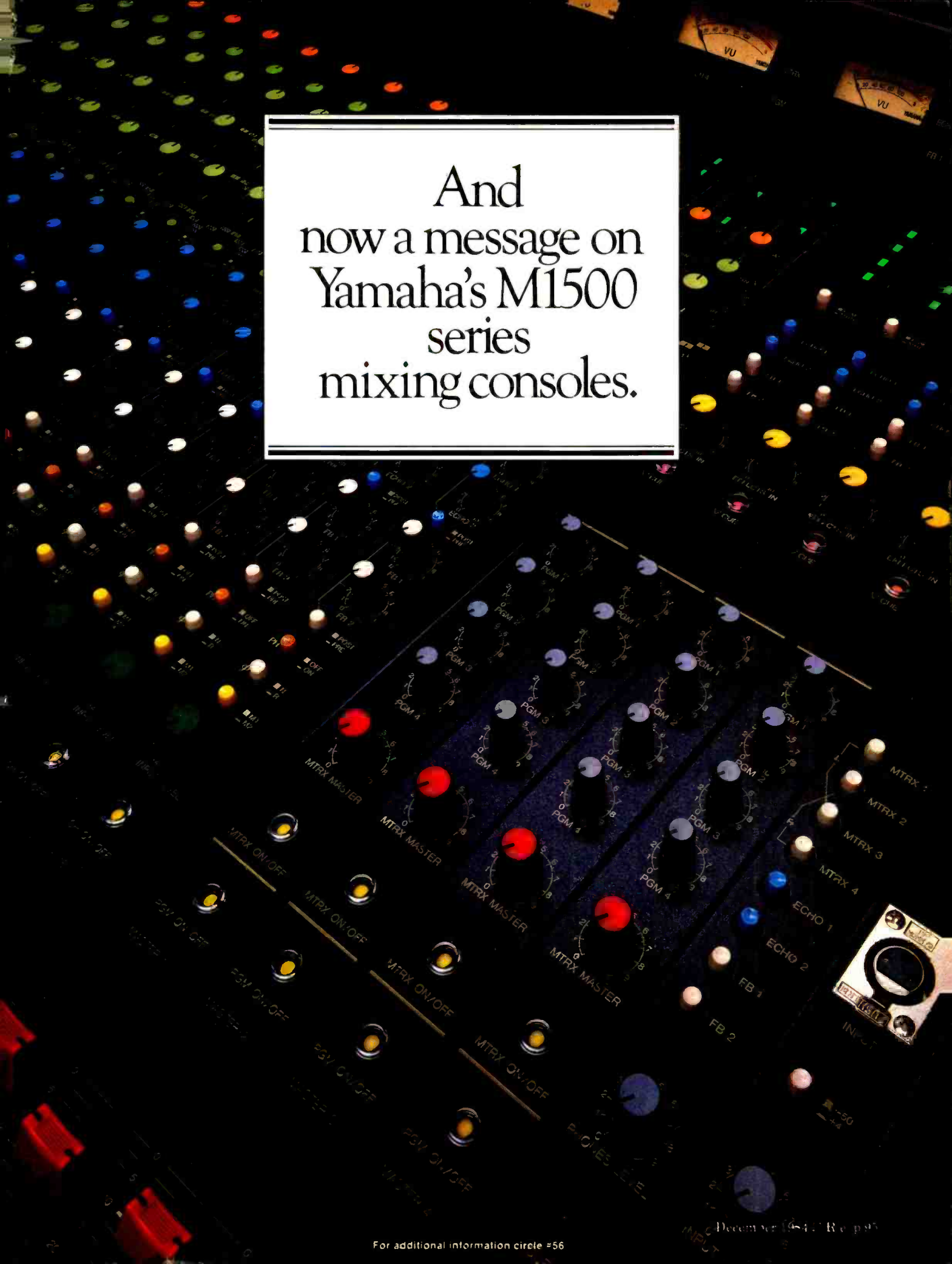
To date, two additional multichannel evaluative recordings have been made involving a total of 27 different microphones (all in stereo) for the three tests. Every effort was made to duplicate in the second and third sessions the conditions of the first. Soprano Carol Meyer again sang "Come, My Love" by Mozart, and "Summertime" by Gershwin, the songs auditioned most often in our first evaluation. She and the Steinway nine-foot concert grand piano were positioned in exactly the same locations as before on the Clapp Recital Hall stage; the microphones were as similarly positioned as possible (depending upon stereo technique); the same control room equipment was used; the same brand and type of magnetic tape, etc. However, we did utilize the services of a different accompanist, Patricia Calahan, since the first pianist was unavailable.

As a control measure, the microphones receiving the highest composite "ratings" in the first evaluation (Neumann SM69fet and TLM170) were included in the second evaluation. In the third evaluation, the TLM170 pair was retained and a Sixties-vintage SM69 (tube) stereo microphone was substituted for the FET version. These particular Neumann microphones belong to the University's Recording Studios; all other units were on loan, as noted above.

Tables 1 and 2 list the microphones now under consideration, with pertinent information about the conditions of the recording sessions. As before, no low-frequency roll off or output attenuation switches were activated; no microphones were modified or tampered with in any way; and no units were used in a manner that did not conform to the intentions of the manufacturer. The dates of manufacture were within the last one or two years (1982-84) with the exceptions of the SM69 tube model (1968) and the SM69fet (1971).

Microphones Under Study

All of the units covered in this arti-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now a message on
Yamaha's M1500
series
mixing consoles.

M1516A

GENERAL SPECIFICATIONS

FREQUENCY RESPONSE +0, −3dB, 20Hz to 20kHz; +0, −0.5dB, 30Hz to 15kHz.

TOTAL HARMONIC DISTORTION (THD)*
Less than 0.5% @ + 10dB, 20Hz to 20kHz. Less than 0.1% @ + 20dB, 50Hz to 20kHz.

HUM AND NOISE* (20Hz to 20kHz, 150Ω source, Input Selector set at "−60")
- −128dBm Equivalent Input Noise (EIN);
- −95dB residual output noise with all Faders down.
- −73dB PROGRAM OUT (77dB S/N); Master Fader at nominal level & all Input Faders down.
- −64dB PROGRAM OUT (68dB S/N); Master Fader and one Input Fader at nominal level.
- −73dB MATRIX OUT; Matrix Mix and Master controls at maximum, one PGM Master Fader at nominal level, and all Input Faders down.
- −64dB MATRIX OUT (68dB S/N); Matrix Mix and Master controls at maximum, one PGM Master Fader and one Input Fader at nominal level.
- −70dB FB or ECHO OUT; Master level control at nominal level and all FB or ECHO mix controls at minimum level. (Pre/Post Sw. @ PRE.)
- −64dB FB or ECHO OUT (68dB S/N); Master level control and one FB or ECHO mix control at nominal level. (Pre/Post Sw. @ PRE.)

MAXIMUM VOLTAGE GAIN (Input Selectors set at "−60" where applicable)
PROGRAM & MATRIX 84dB; Channel In to the corresponding output. EFFECTS 20dB; Effects In to PGM Out.
FB & ECHO 94dB; Channel In to FB/ECHO Out. SUB IN 10dB; Sub In to PGM Out.

EQUALIZATION (±15dB maximum)
LOW: 50, 100, 200, 350, 500Hz, shelving. HIGH MID: 1.2, 2, 3.5, 5.7kHz, peaking.
LOW MID: 250, 350, 500, 700, 1000Hz, peaking. HIGH: 10kHz, shelving.

HIGH PASS FILTER 18dB/octave rolloff below 80Hz.

PHANTOM POWER For remote powering of condenser microphones, +40V DC can be switched on via a rear panel Master phantom power switch. When an individual Input Phantom switch is also On, voltage is applied to pins 2 and 3 of that input's balanced XLR connector.

DIMENSIONS/WEIGHT M1516A 34" W x 36 1/2" D x 14 1/2" H 147 lbs. M1524 55 3/4" W x 36 3/4" D x 14 1/2" H 213 lbs. M1532 55 3/4" W x 36 3/4" D x 14 1/2" H 231 lbs.

*Measured with a 6dB/octave filter @ 12.47kHz; equivalent to a 20kHz filter with infinite dB/octave attenuation.

The specs shown are for the 16-channel M1516A console. When you need the same outstanding performance but more channels, there's the 24-channel M1524 and the 32-channel M1532. All three mixers have remote rack-mounted power supplies and are ideal for just about any fixed or portable sound reinforcement or broadcast application.

Of course, all three M1500 consoles have legendary Yamaha quality, reliability and craftsmanship. Which explains why you see Yamaha mixers wherever you look. Studios. Concert halls. Clubs. Theatres. Churches. We could go on, but you get the message.

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YAMAHA

For additional information circle #57

MICROPHONE ASSESSMENTS

cle are condenser microphones. The B&K 4007 and Crown PZM-31S are electrets; all others require DC polarizing voltages across the condenser elements.

SESSION #2:

• The **AKG C34** is a stereo microphone employing two 18mm (0.7 inch) twin capsules based on the well-known CK1 design. One of the capsule housings may be rotated through a 180-degree arc to permit use of the MS or XY coincident stereo techniques. The C34 operates over a wide range of phantom-power voltages (9 to 52 volts, based on DIN 45 596)[1] and offers nine remotely selectable patterns per capsule. This microphone is physically smaller than the companion AKG C422 large-diaphragm stereo version, yet both provide essentially the same operating features.

• The **AKG C414EB** is a large-diaphragm (one-inch diameter) studio microphone with four selectable patterns: omnidirectional, cardioid, hypercardioid, and figure-of-eight. It is a descendant of the famous C12 tube model, which has recently been re-engineered and introduced again as "The Tube." Like the other FET designs from AKG, the C414EB operates on phantom powering. Standard features include 0, -10, and -20 dB pre-attenuation and "flat," 75 Hz, and 150 Hz low-frequency attenuators. The C414 Series, one of the most widely used of all condenser studio microphones, is still evolving: the C414EB/P48, recently introduced but unavailable for this evaluation, is claimed by

TABLE 1: SPECIFICATIONS OF TWELVE MICROPHONES USED DURING SECOND STEREO EVALUATION SESSION

Recording session of May 9, 1984 (class project for Music 25:214, Recording Techniques, Spring semester 1984).
Concert hall temperature: 68° F (20° C); ±3° F.
Relative humidity: between 45 and 55%.
Barometric pressure: 29.68 inches of mercury (753.9 mm Hg, 100.5 kPa).
Nominal microphone input impedance of Neve 5315/24 console: 1.2 kohms (transformer; balanced and floating).
Nominal line input impedance of console: 10 kohms (transformer; balanced and floating).

Brand and Model Numbers	Serial Numbers	Source Impedance	Recommended Load Impedance	Console Gain (approx.)
1. AKG C34	208	200 ohms, nominal; ±20%	500 ohms or greater	50 dB
2. AKG C414EB	10306, 10323	150 ohms or less	500 ohms or greater	45 dB
3. AKG C460B	CK1 Capsules: 71038, 71052 Amplifiers: 04149, 03256	120 ohms or less	500 ohms or greater	45 dB
4. Bruel & Kjaer 4007	104009, 1040039	Less than 30 ohms [actual]	Unspecified	55 dB
5. Calrec CM2050C	Capsules: 3445, 3448 Amplifiers: 5464, 5505	200 ohms rated	500 ohms [or greater]	50 dB
6. Calrec MkIV Soundfield	625	100 ohms approx.; unbalanced, line level ("0 dBu"), transformerless	600 ohms, minimum	0 dB [total gain pre-amplifier & control unit: 50 dB]
7. Neumann KMF4	Capsules: 030, 046 Amplifiers: 082, 081	150 ohms, nominal	750 ohms, minimum 1,000 ohms, nominal	40 dB
8. Neumann KU81	02/038	150 ohms, nominal	750 ohms or greater, nominal	40 dB
9. Neumann SM69fet	2094, mfr. date: 1971	200 ohms, nominal	1,000 ohms, minimum	40 dB*
10. Neumann TLM170	3/100, 4/100	150 ohms, nominal, transformerless	1,000 ohms, minimum	40 dB
11. Schoeps CMC54U	Capsules: 29327, 29339 Amplifiers: 10174, 10173	Approx. 35 ohms [actual], transformerless	600 ohms or greater	35 dB
12. Schoeps MSTC54 "ORTF"	Capsules: 29101, 29108 Amplifier body: 246	Approx. 40 ohms [actual], transformerless	600 ohms or greater	35 dB

*Note: the value of 35 dB given in the April 1984 *R-e/p* article is incorrect; the true value is *40* dB for the SM69fet.

MICROPHONE ASSESSMENTS

AKG to have "the highest dynamic range of any microphone we know."[2]

• The **AKG C460B** "combo" Series is an updated, lower-noise version of the C451/C452EB Series (still available) reported upon previously. The C460B units evaluated here were supplied with CK1 fixed-cardioid 18mm capsules. The amplifier sections have 0 and -10 dB preattenuation, plus "flat," 70 Hz, and 150 Hz bass roll off characteristics. The AKG "combo" modular system is exceptionally versatile, offering a comprehensive choice of capsule types and operating features. All of the microphones loaned by AKG for these tests were well documented, with individual on-axis and off-axis (180-degree) frequency response test results for the specific patterns appropriate for each capsule.

• **Bruel & Kjaer** microphones have enjoyed a long and well-deserved reputation as the standard measurement devices for acoustical testing and research applications. Currently, the 4000 Series of microphones is being marketed for use in concert and studio recording. Since extensive calibration data is supplied with each microphone (on-axis free-field response, sensitivity in mV/Pa, dynamic range and noise information, etc.), the purchaser has available a test and measurement microphone as well as a high-quality unit suitable for studio use. The Model 4007 used in this experiment is the small-diameter (12mm — half-inch — capsule), 48-volt phantom-power version; companion units offer 16mm capsules and high-voltage power supply options. B&K is committed exclusively to the manufacturing of pressure transducer (omnidirectional) microphones for audio recording.[3]

• The **Calrec CM2050C** is one member of a family of general-purpose microphones designed for recording, broadcast, sound reinforcement, and film applications, similar to the modular series from AKG, Neumann, Schoeps, and other manufacturers. The 20mm (0.8-inch) cardioid capsule of the CM2050C is detachable, and may be replaced with omnidirectional or bass roll off cardioid capsules, the latter available with or without windscreens.

• The **Calrec MkIV Soundfield** concept is truly exceptional: no other system offers the engineer as much in terms of overall versatility, true three-dimensional recording capability, and post-production manipulation. Although it is both complex and expensive (almost $5,000), the Soundfield system is an innovative accomplishment in microphone design. The single microphone contains four capsules mounted as surfaces of a regular tetrahedron, oriented symmetrically and "looking" into three-dimensional space. The physical and electrical properties of the capsule array approximate those of a model *spherical* transducer as it responds to variations in sound pressure levels. The resulting applications of the design include use of *any* of the known coincident two-channel stereo techniques, as well as Ambisonic single-point multichannel capabilities. Furthermore, since horizontal *and* vertical information is included in the Soundfield technique, true three-dimensional or "periphonic" surround-sound reproduction is now possible from a single microphone.

The post-production opportunities provided by the Soundfield system are especially attractive. Even if only three channels of direct microphone outputs are recorded (pre-matrix or "B-Format"), the engineer can "steer" and "zoom" the final stereo image over a wide range of possibilities, while at all times preserving a mono-compatible signal. These opportunities are available from the MkIV Control Unit, which has not only compreh-

TABLE 2: SPECIFICATIONS OF SIX MICROPHONES USED DURING THIRD EVALUATION SESSION

Recording session of June 13, 1984*.
Concert hall temperature: 68° F (20°); ±3° F.
Relative humidity: between 50 and 60%.
Barometric pressure: 30.05 inches of mercury (763.3 mm Hg 101.7 kPa).
Nominal console input impedances: see Table 1.

Brand and Model Numbers	Serial Numbers	Source Impedance	Recommended Load Impedance	Console Gain (approx.)
1. Crown PZM-31S	PZM units: 003608, 003615 Interface units: 015649, 015710	150 ohms [actual]	1,000 ohms or greater	50 dB
2. Milab DC-96B	4931, 5020	200 ohms, nominal	1,000 ohms, nominal	45 dB
3. Milab LC-25	7672, 7902	300 ohms, each signal lead to ground, transformerless; 200 ohms, nominal equivalent	1,000 ohms, nominal (balanced and floating input)	45 dB
4. Milab XY-82	6334	200 ohms, nominal	1,000 ohms, nominal	40 dB
5. Neumann SM69 (tube)	1601, mfr. date: 1968	200 ohms, nominal	1,000 ohms, minimum	40 dB
6. Neumann TLM170**	3/100, 4/100	150 ohms, nominal, transformerless	1,000 ohms, minimum	40 dB

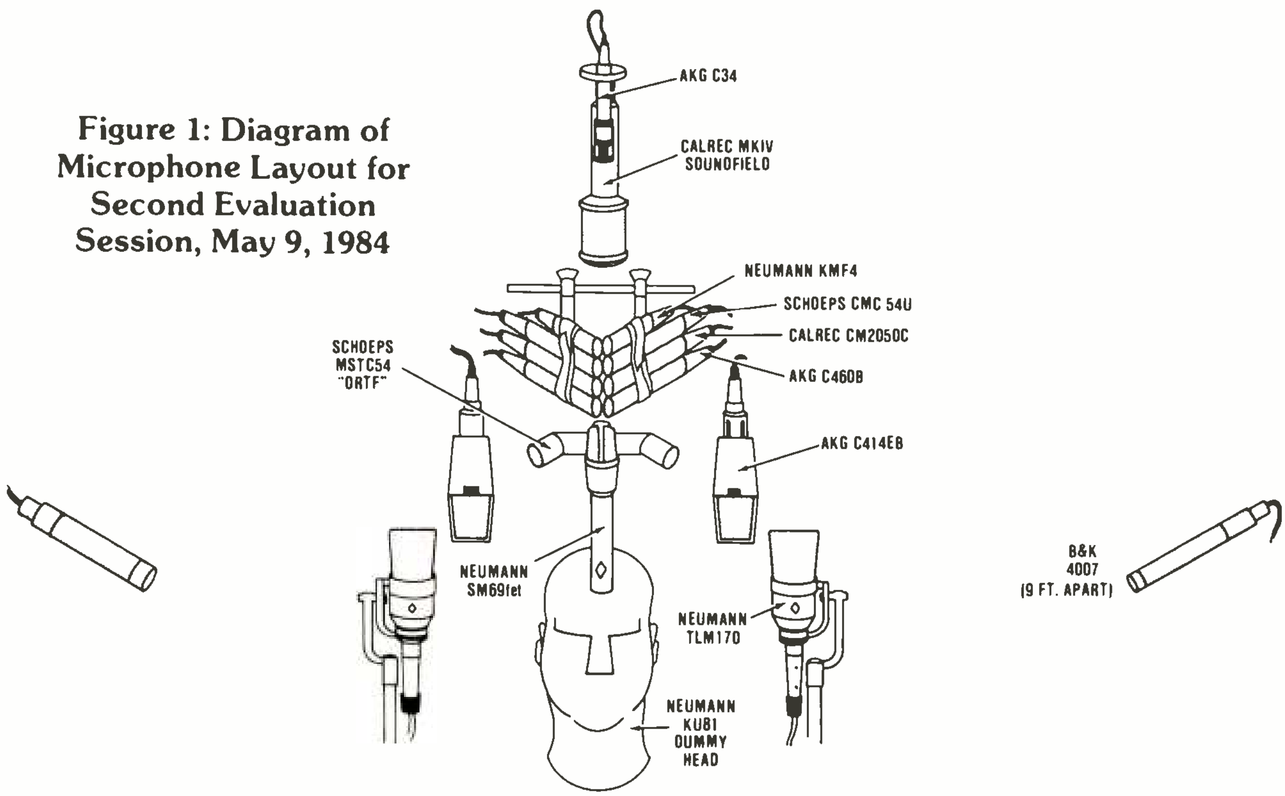
*Private recording session: only performers, photographers, and engineering staff present.
**Note: These are the same units as no. 10 in Table 1 above, and no. 6 in first evaluation.

ensive control, metering, and test-tone features, but also line-level inputs for manipulation of pre-recorded "B-Format" information. (The four "B-Format" signals X, W, Y, and Z are front/back-facing figure-of-eight, omnidirectional, left/right, and up/down, respectively; for horizontal sound fields, the Z signal is set to zero.) No other microphone in this series of evaluations elicted as much interest from our musicians, engineering staff, and students.[4,5] [For more details the interested reader is referred to the article "Ambisonic Surround-Sound Technology for Recording and Broadcast," published in the December 1983 issue of *R-e/p — Editor.*]

• The **Neumann KMF4** is the latest and smallest in the KM Series of miniature multipurpose microphones. It employs a cardioid capsule in a housing measuring only 17mm in diameter by 38mm (1.5 inches) long, which may be located up to 5m (16 feet) from the KM amplifier. The manufacturer's intention was to provide an easily concealed, cardioid studio microphone for stage and television productions, permitting use at greater distances than those usually associated with miniature pressure-transducer (omnidirectional) units.

• The **Neumann KU81** is the redesigned successor to the KU80 dummy head. The dummy head technique (referred to more artfully in German as *Kunstkopf*) is *binaural*, or intended for two-channel headphone reproduction. However, the design criteria for the KU81 have taken into account its applications for eventual stereophonic loudspeaker reproduction (as well as binaural headphone monitoring) by equalizing the system outputs for uni-

Figure 1: Diagram of Microphone Layout for Second Evaluation Session, May 9, 1984



Actual microphone layout on Clapp Recital Hall Stage during second evaluation session, May 9, 1984

form frequency response in a diffuse, or reverberant, sound field. [6,7,8] One obvious reason for developing two-channel stereo techniques that are headphone/loudspeaker compatible is today's widespread use of "Walk-person"-type listening devices.

• The **Neumann SM69fet** and **TLM170** are respectively the firm's coincident stereo (MS, XY) and transformerless studio microphones. (See full descriptions on page 123 of the April 1984 issue of *R-e/p*.)

• The **Schoeps CMC54U** is the "flat" response, cardioid member of the Colette family of modular microphones. Like the AKG C450/460 "combo" and Neumann KM Series, the Schoeps line offers a comprehensive set of choices. Distinctive features of the Colette microphones are transformerless, high-level outputs, and the exclusive use of small diameter (approximately 20mm) *single* capsules — even for hypercardioid and bidirectional patterns. Selection of polar characteristics in multipattern Schoeps microphones is accomplished by changing acoustic-mechanical elements, and not by varying polarization voltages, as in the case of dual-element capsules. A recent addition to this series is the BLM3 boundary layer microphone, identical *in concept* to the Crown PZM-31S units borrowed for this evaluation.

• The **Schoeps MSTC54** is a stereo microphone designed according to the ORTF principle of recording. This technique, developed by the French national broadcasting authority, Office de la Radiodiffusion-Television Francaise, utilizes two cardioid pickups spaced 170mm (6.7 inches) apart and placed at a 110-degree angle. Since its MK 4 capsules are identical to those in the CMC54U microphones, the availability of this system offered the opportunity to compare the ORTF and XY techniques. The MSTC54 is the only stereo microphone system known to this author to be constructed according to ORTF principles. Most persons devoted to this technique simply set up two separate cardioid microphones at the proper spacing and angle. Schoeps has available the UMS20 universal stereo bracket, which easily accomplishes this purpose, in addition to providing mounting configurations for XY, MS, and near-coincident methods for recording in two-channel stereo.

SESSION #3:

• The **Crown PZM-31S** is an example of Pressure Zone™, boundary-layer, or acoustical-boundary microphone design. This principle "seeks to integrate the transducer into one of the natural planes found in the acoustical environment to be reproduced. Embedding the microphone in an acoustically reflective surface where only pressure phenomena exist . . . eliminates the comb-filter and interference effects usually considered unavoidable in more conventional setups, where complex cross-patterns of standing waves cause frequency-dependent cancellations and reinforcements throughout the spectrum."[9]

These advantages must be balanced against known problems of the boundary-layer concept. Low-frequency response is directly dependent upon the area of the acoustical boundary: The transducer is usually mounted on floor, wall, or other very large surfaces, which should approximate one-half of the wavelength of the lowest frequencies to be recorded. Furthermore, the hemispherical polar pattern of PZM™ microphones behaves very much like an omnidirectional pattern, greatly limiting the effective use of such devices in stereo recording. "PZMs used as a spaced pair for stereo recording have all the flaws that spaced omnis do, such as poor image focusing. PZMs provide sharper imaging if mounted back-to-back on boundaries as a coincident or near-coincident pair."[10] Unfortunately, such boundaries are likely to become ungainly and cumbersome if they are expected to provide adequate low-frequency response *and* mounting positions conducive to proper coincident stereo techniques.[11]

• The **Milab DC-96B** is unusual for at least two reasons. First, it utilizes a fixed-cardioid, dual-membrane capsule (open on both sides) that obtains its pattern from polarizing voltages. This can be considered the opposite approach from that employed by Schoeps, where single capsules for all patterns are the rule. The second unusual feature of the DC-96B capsule is the rectangular shape of the aluminized membranes. The apparent design premise of the rectangular capsule relates to the manufacturer's preference for two minor resonances (at the wavelengths of the two dimensions of the rectangle) in comparison to a single, more pronounced resonance determined by the diameter of a circular capsule. However, there are circular-capsule Milab designs, as seen in the following two examples.

• While the **Milab LC-25** may look like a vocalist's microphone for hand-held use, this was not the manufacturer's design intention; instead it is a fixed-cardioid, large-diaphragm studio microphone for use three to six feet or more from the sound source. Applications of this transformerless unit range from use with solo instruments or voices to the recording of large groups in reverberant environments. It can be supplied as a balanced, transformerless, line-level output device, in which case it is referred to as the LC-25LL.

• The **Milab XY-82** is a small-diaphragm, fixed-cardioid stereo microphone intended for use in the XY coincident technique (the MS technique being impossible with this design because no bidirectional pattern is provided). One of the capsule housings may be rotated through a 180-degree arc, permitting coincident pickups at varying angles, or use where both capsules are in the same plane (i.e., parallel), as in certain "backup" or multiple-feed situations.

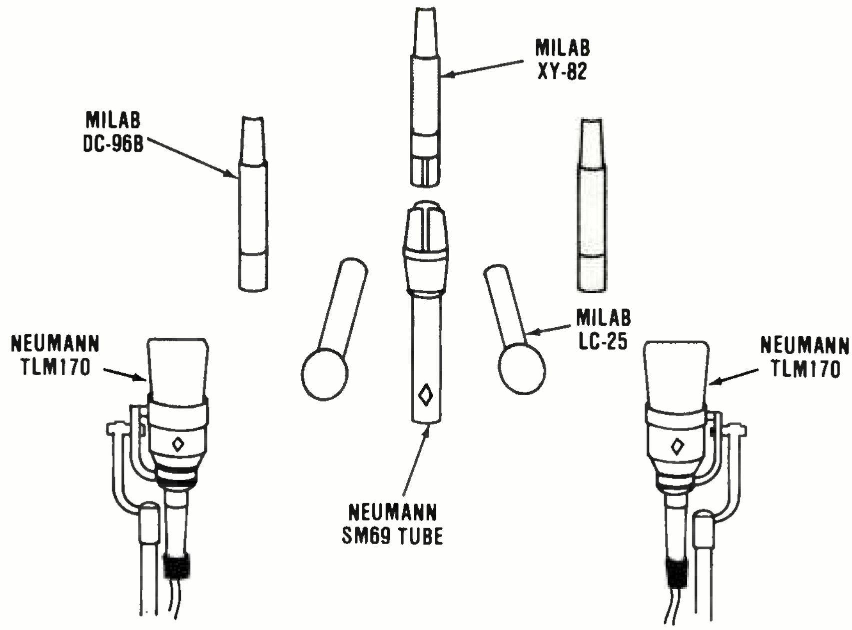
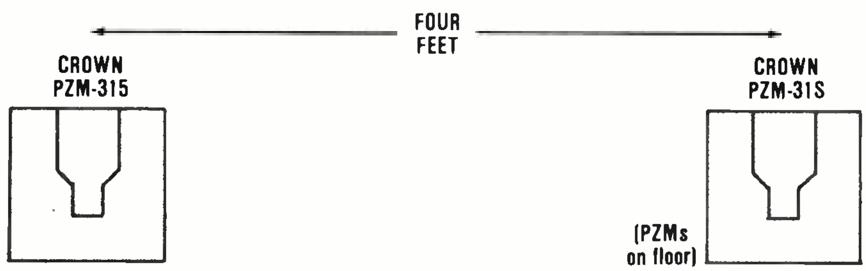


Figure 2: Diagram of Microphone Layout for Third Evaluation Session, June 13, 1984.



The XY-82 can be considered a direct competitor to the AKG C33 stereo microphone, which is identical in appearance to the more versatile C34 stereo unit of this evaluation, but which is limited to two fixed-cardioid patterns for XY use. All Milab units loaned for this experiment were documented with individual frequency response curves for each capsule.

• The **Neumann SM69 (tube) stereo microphone** is the immediate predecessor of the current FET version, utilizing two miniature AC701 vacuum tubes as impedance-matching devices (one each for the dual-membrane 28mm [1.1-inch] capsules) plus a special high-voltage power supply and remote pattern selector (nine patterns per capsule). In all other respects regarding operating features and physical appearance of the microphone itself, it is identical to the contemporary FET version. (Incidentally, The Neumann emblem is black on vacuum-tube models, and purple on field-effect transistor models.)

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

Stereo Microphone Techniques

A brief overview of the "classical" microphone techniques for recording in two-channel stereo was provided in the previous survey, and is a topic that has received an enormous amount of attention throughout the literature; it continues to be a source of interest — even fascination — among audio researchers and practitioners. A lengthy bibliography could be compiled on the subject: items 12 through 18 in the notes and references listed at the end of this article list some selected sources, spanning a 26-year period.

This second round of microphone evaluations was conceived not only to provide a means for comparing perceptions of qualitative differences of many brands and models of microphones, but also to document and compare a variety of stereo microphone techniques. There is no doubt that subjective judgements, on my own part and on the part of the many

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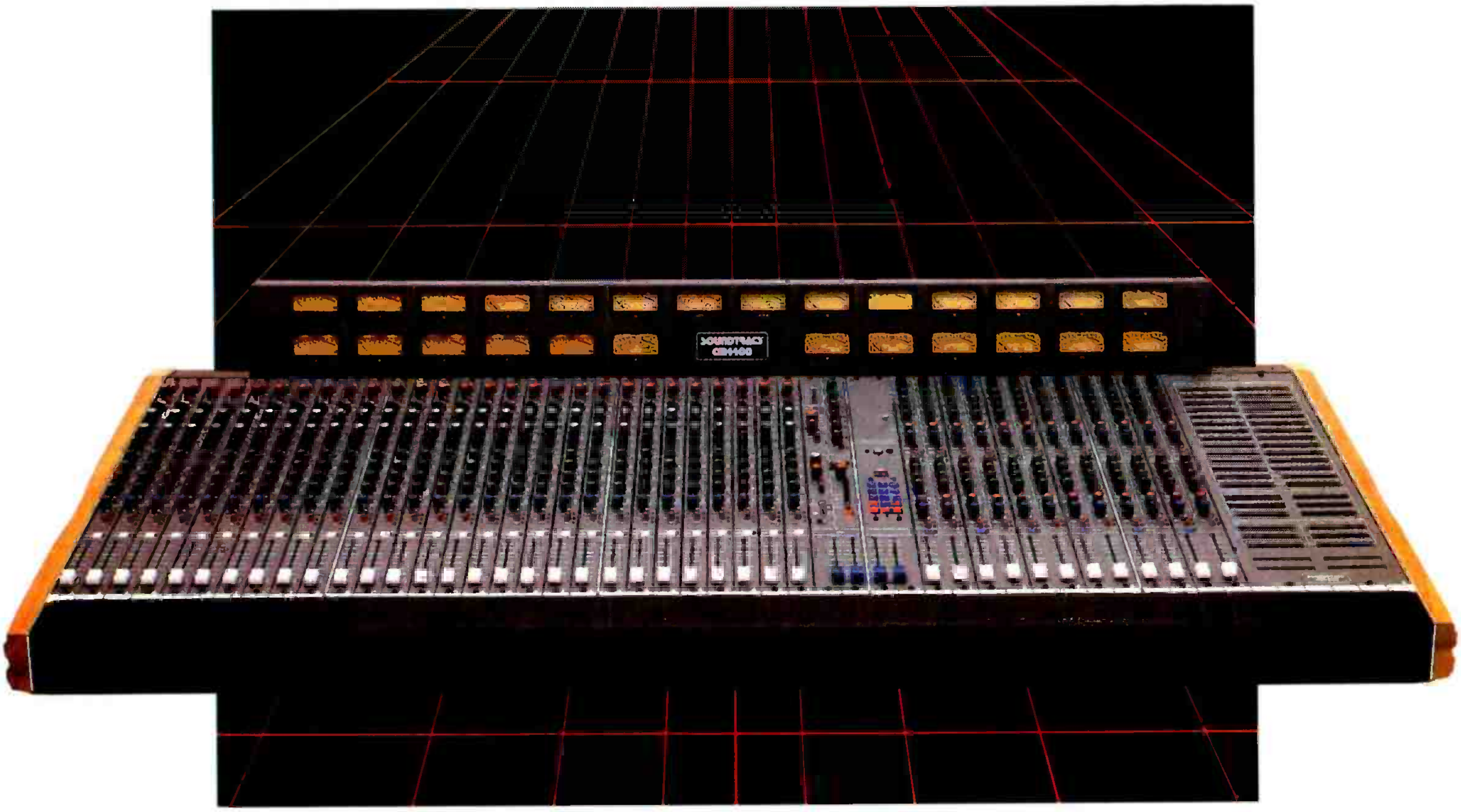

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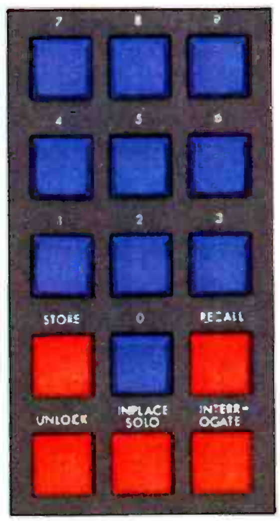

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For additional information circle #61

other participants in the listening tests, could not and cannot be avoided in evaluations such as this. Subjective elements, it must be granted, often make it difficult to separate *microphone* qualities from *stereo* qualities. But I maintain that there is no substitute for direct experience, under actual, practical recording conditions, in learning firsthand about the inherent differences, advantages, and disadvantages of various stereo recording techniques, or about the microphones themselves that are required for the realization of such techniques.

Table 3 lists the stereo techniques employed in the second and third evaluations with the appropriate microphones as utilized. Note that each pair of microphones, and each stereo microphone, was set up in Clapp Recital Hall so that all received as identical an acoustical signal *as possible*. Within the confines of the respective stereo techniques, no microphones were intentionally positioned to have an advantage over the others. Again, I readily grant that it is physically impossible to set up this number of microphones in such a way that each pair, or each stereo unit, receives *exactly* the same acoustical information.

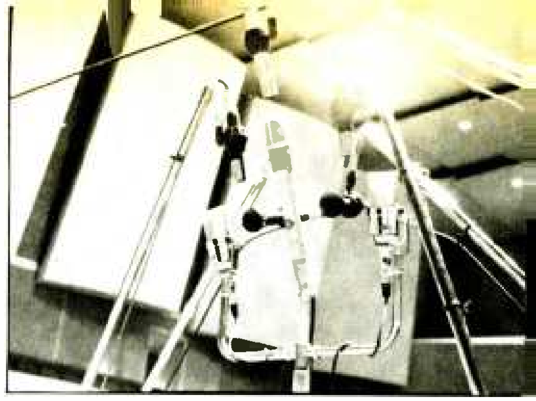
Stereo Monitoring

From the outset of our listening tests, it became evident that we could not expect our 20 participating auditioners in the second and third round of evaluations to concentrate simultaneously upon their perceptions of *stereo* quality and *microphone* quality. Hence, the composite "ratings" listed later in the Microphone Evaluations section were based upon a conscious emphasis on the qualities of the individual transducers themselves; judgements relating to variations in stereo-technique preferences, however, could not be completely avoided.

The rankings of stereo technique that follow are therefore entirely my own, and have been formulated after extensive listening over our Control Studio A monitor loudspeakers (two each JBL Model 4320 for stereo and four each Model 4310 for surround sound) *and* a selected pair of AKG K-141 stereo headphones.[19] In all listening tests, the loudspeakers and headphones were driven by the appropriate outputs of McIntosh MC2505 two-channel power amplifiers.

The frequency response of the AKG K-141 headphones has been measured at the University's Speech and Hearing Center utilizing a Bruel & Kjaer "KEMAR" head containing a B&K half-inch, Model 4134 calibrated pressure transducer within the artificial ear. The K-141 headphones were driven by the headphone output of a Crown D75 power amplifier, which was in turn fed from a General Radio 1304-B oscillator calibrated to remain within ±0.5 from 10 Hz to 25 kHz. The output of the B&K 4134 pressure microphone was connected to a B&K Model 2033 spectrum analyzer, the paper-tape linear response curve from which was then redrawn on semi-log paper by Steven Sergeant, who supervised the test.

Figure 3 shows the response in the left K-141 earphone. This curve, which resembles but is not identical to that of the right earphone, is characteristic of "artificial ear" measurements of earphone response. Our measurement confirmed that this specific K-141 conforms to the design expectations of the manufacturer.[20]



Detail of microphone layout during third evaluation session, June 13, 1984.

TABLE 3: STEREO MICROPHONE TECHNIQUES UTILIZED DURING CURRENT EVALUATIONS

Stereo Technique*	Microphones	Evaluation Number (2) = May 9, 1984 (3) = June 13, 1984
Coincident MS, Matrixed to Left and Right Channels	AKG C34	(2)
	Calrec MkIV Soundfield**	(2)
	Neumann SM69fet	(2)
	Neumann SM69 (tube)	(3)
Coincident Cardioid XY	AKG 460B	(2)
	Calrec CM2050C	(2)
	Calrec MkIV Soundfield**	(2)
	Neumann KMF4	(2)
	Schoeps CMC54U	(2)
	Milab XY-82	(3)
Coincident Figure-of-Eight (Bidirectional) or Blumlein XY	Calrec MkIV Soundfield**	(2)
Slightly Spaced (1 ft., 30cm or less) Cardioid Pairs, approx. 45° toed-out angle	AKG C414EB	(2)
	Neumann TLM170	(2) *and* (3)
	Milab DC-96B	(3)
	Milab LC-25	(3)
ORTF	Schoeps MSTC54	(2)
Widely-spaced (9 feet/2.8m) Omnidirectional Pair	Bruel & Kjaer 4007	(2)
Widely-Spaced (4 feet/1.2m) PZM pair, on floor	Crown PZM-31S	(3)
Binaural Dummy Head	Neumann KU81	(2)
Single-point Ambisonic Surround-sound	Calrec MkIV Soundfield**	(2)

*Note: All microphones were located approximately 7 to 8 feet (2 to 2.5m) above the floor of the stage on a line about 4 feet (1.2m) downstage from the singer, *except* the B&K 4007s and the PZM units. The 4007s were on 6 feet (1.8m) stands, on the same line as the cluster of microphones, spaced 9 feet or 2.8m [!] apart. The PZMs were on the floor, on a line approximately 3 feet (90cm) from the singer, spaced 4 feet or 1.2m apart. See photographs of the recording sessions, and Figures 1 and 2.

**Note: In addition to recording the *stereo* output from the Soundfield system onto the Studer/telcom 15 ips 24-track tape during the May 9 evaluation, its B-Format outputs were simultaneously recorded by a four-channel Ampex ATR-100, operating with Dolby A-Type noise reduction systems (15 ips, Ampex 456 Grand Master half-inch tape). The Ampex/Dolby B-Format recording provides the recovery of all four techniques (**) shown above (Matrixed MS, Cardioid XY, Blumlein XY, and Single-point Ambisonic), as well as others beyond the scope of this project. I am indebted to Steven Sergeant, our audio engineer, for his assistance with the Soundfield B-Format recording. □□□

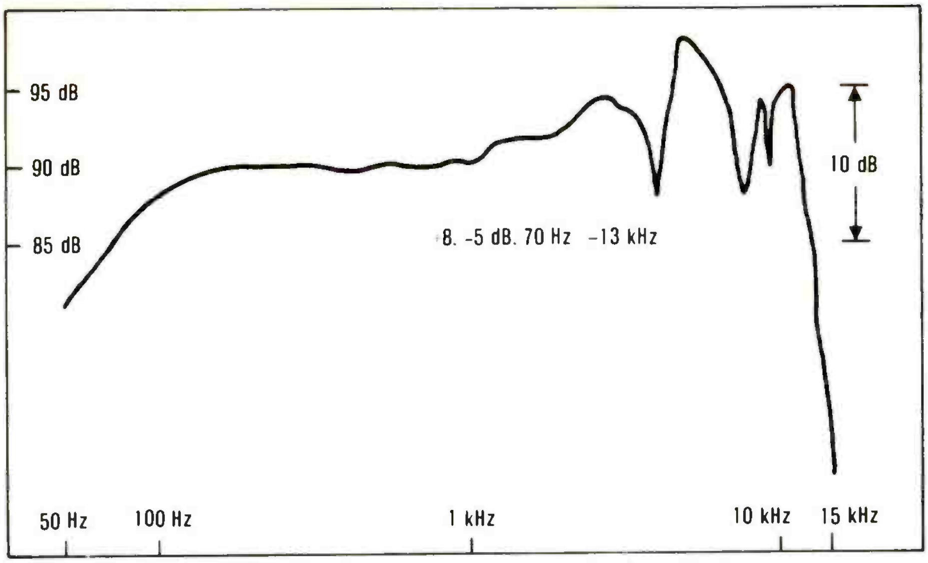


Figure 3: Left earphone response of selected pair of AKG K-141 stereo headphones.

Notes and References

1. All microphones involved in the second and third experiments are powered according to DIN 45 596, except the MkIV Soundfield and the two versions of the SM69 — these use special power supply/control units offering remotely selectable patterns. The AKG C34's pattern controller works in conjunction with standard DIN 45 596 power sources.

2. This claim has not gone unchallenged; see "Editorial note." *Journal of the Audio Engineering Society*, Vol. 32, No. 9 (Sept. 1984), page 678.

3. See Philip White, "Evaluation of studio microphone performance using time delay spectrometry techniques." Paper, Audio Engineering Society Convention, Oct. 23-27, 1982. Naerum, Denmark, Bruel & Kjaer, [1983].

4. Michael A. Gerzon, "With-height sound reproduction." *Journal of the Audio Engineering Society*, Vol. 21, No. 1 (Jan.-Feb. 1973), pages 2-10.

5. J. Howard Smith, "The Sound Field microphone." *db*, Vol. 12, No. 7 (July 1978), pages 34-36.

6. Georg Neumann GmbH, specification sheet no. 12212 80201 for KU81 dummy head. Berlin, August 1982.

7. Klaus Genuit, "A contribution to the optimizing of some 'dummy head' recording systems." Report of the 12th Tonmeister Conference, Munich, 1981, pages 218-243 (in German).

8. Stephen Peus, "Development of a new studio artificial head" (transl. S.F. Temmer). *db*, Vol. 18, No. 5 (June 1984), pages 34-36.

9. Schalltechnik Dr.-Ing. Schoeps GmbH, specification sheet no. 830502 for BLM3 boundary layer microphone. Karlsruhe, 1983.

10. Bruce Bartlett, letter to the author, May 22, 1984.

11. See Michael E. Lamm and John C. Lehmann, "Realistic stereo miking for classical recording." *Recording Engineer/Producer*, Vol. 14, No. 4 (Aug. 1983), pages 98-109.

12. *Journal of the Audio Engineering Society*, Vol. 6, No. 2 (Apr. 1958). Issue devoted to stereophonic sound; includes a reprint of Alan Dower Blumlein's British Patent Specification 394,325, pages 91-98, 130.

13. Carl Ceoen, "Comparative stereophonic listening tests." *Journal of the Audio Engineering Society*. Vol. 20, No. 1 (Jan. - Feb. 1972), pages 19-27.

14. Bruce Bartlett, "Stereo microphone technique." *db*, Vol. 13, No. 12 (Dec. 1979), pages 34ff.

15. John Eargle, *The Microphone Handbook*. Plainview, NY, Elar Publishing, 1981.

16. Wesley L. Dooley and Ronald D. Streicher, "M-S stereo: a powerful technique for working in stereo." *Journal of the Audio Engineering Society*, Vol. 30, No. 10 (Oct. 1982), pages 707-718.

17. Lamm and Lehmann. *op. cit.*

18. Michael Dickreiter, *Mikrofon-Aufnahmetechnik* (*Microphone recording technique*). Stuttgart, S. Hirzel Verlag, 1984 (in German).

19. See the previous article (*R-e/p*, April 1984, page 126) for an anechoic frequency response test of the JBL 4320s. Information regarding anechoic test results for the JBL 4310s may be obtained by writing to Lowell Cross, School of Music, The Univ. of Iowa, Iowa City, IA 52242.

20. Internal memorandum, Akustische and Kino-Gerate GmbH. Dr. C. Poldy, "The philosophy of modern headphone design." AKG Information [in-house paper], Vienna, undated. The latter describes the problems of measuring headphone response with an artificial ear.

In the conclusion of this two-part article, to be published in the February 1985 issue, Lowell Cross will present a subjective appraisal of the various stereo microphone techniques outlined above, and will tabulate the results of listening tests made during the two evaluation sessions — *Editor*.

STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

Northeast:

□**MILLBROOK SOUND STUDIOS** (Millbrook, New York) is a new 24-track facility run by engineer **George Cowan** and studio manager **Rick Kennell**. Featured is a new custom-built 36-input **Neotek Series 3C** console with 36 inputs and eight stereo sub-groups. According to Cowan, "The console was chosen primarily for its unsurpassed audio quality. Neotek's design philosophy coincides with our own outlook. The audio circuitry, beginning with the transformerless mike pre-amps, has the dynamic range necessary for digital recording. The channel equalizers are wonderfully musical; it is rare to have to use any outboard EQ for anything now. Though the board is logical and very user-friendly, the emphasis is on good sound. We wanted automation but not at the price of mediocre sound quality. The Neotek design, with its sub-grouping and group muting capabilities, allowed us to wait for a really good automation system. We are looking toward a versatile automation system, like moving fader GML, with its ability to read automation data generated by different systems." The studio boasts a large 25- by 40-foot relatively quiet room, a large very "live" room, and two isolation booths. Other control-room equipment includes an **Ampex MM-1200** 24-track, and autolocator, **Otari MTR-10** and **MTR-12** half-inch, two **Orban Dynamic Siblance Controllers**, two **Aphex Aural Exciters**, **Valley People Kepex IIs** and **Gain Brain IIs**, a **UREI 1178** dual peak limiter, two **dbx Model 160X** limiters/compressors, four **Orban** stereo parametric EQs, four **White** third-octave EQ, an **Eventide H910 Harmonizer**, an **Ursa Major 8X32** digital reverb, a pair of **UREI 811Bs**, **JBL 4411s**, **Yamaha NS-10Ms**, and **Auraton Sound Cubes**, and a wide range of microphones from **AKG**, **Sennheiser**, **E-V**, **Crown**, **Countryman** and **Neumann**. *P.O. Box 317 — Route 44, Millbrook, NY 12545. (914) 667-3733.*

□**CBS RECORDS** (New York City) has taken delivery of two **Sony PCM-3324** digital multitracks. The first phase of a long range plan to equip CBS Records facilities with all-digital capabilities began with the acquisition of **PCM-1610** digital processors. According to **Cal Roberts**, senior VP of operations/marketing, "CBS probably has the largest arsenal of PCM-1610 processors in the country. When we investigated digital multitrack recorders, we looked at a number of different products. No one seemed to have a practical, functioning system like Sony's PCM-3324, which drew a consensus of favorable opinions from all of us here at CBS Records." The 3324's editing capabilities offers tremendous flexibility, believes director of recording operations **Roy Friedman**. "Along with ease of operation, these recorders provide us with the quality we need to meet our rigorous recording standards," he said. "Sony digital recorders are the most reliable. We've had fewer problems with the digital two-track and multitracks than with any other recording technology we've used." Initially the units will be used for mixdowns and remixes of existing classical music releases. Not only will these be made available to consumers on Compact Discs, but CBS Records will take its analog recordings from its archives and transfer them to digital masters to preserve them in their original musical format. Once the archives are remastered, the company plans to begin digitally recording future releases. *New York, NY.*

□**PRODUCTION ONE** (Philadelphia) a new, full-service audio production house recently celebrated its grand opening. "We offer everything a client could need for audio production," says studio **manager Cathy St. John.** "Besides our all new equipment, such as the **BI-MIX 1280** mixing console, **Tascam 58-0B** eight-track and **52** two-track recorders, and **Tascam 122** cassette deck, we have a variety of instruments, including a **Yamaha PF-15** piano, **Roland 106** programmable synthesizer, and a **Linndrum** computer. We have two studios, an isolation booth, pre-production lounge, plus a complete sound effects and music library." **Mike St. John** is the facility's new owner/producer. "This facility has been my dream for a long time," he enthuses. "Now that it's here, my partner **Jeffrey Singer** and I are looking forward to serving the production needs of the area, and will specialize in jingle production and custom TV and radio commercials." *1610 South Second St., Philadelphia, PA 19147. (215) 925-1124.*

□**SQUIRES PRODUCTIONS** (White Plains, NY) has taken delivery of the first **Mitsubishi XE-1 Electronic Editing System** in North America for digital recording of classical music, according to company president **Gregory K. Squires**. The XE-1 editor is designed to work with the two X-80 digital two-tracks recorders the facility also has purchased. "For classical music recording, electronic editing is greatly preferred," Squires explains. "The ability to rehearse your edits before actually performing them allows a degree of perfection unavailable with analog systems. Variable electronic crossfades and the ability to reset levels digitally allow for flawless edits between different takes." Squires has worked on most of the available digital recorders, and his selection of the Mitsubishi XE-1 system was based on an educated, hands-on opinion. "Each machine has its own particular advantages," he says. "But one of the things I respect most about the Mitsubishi X-80 is its ability to monitor during recording, which is a *big* plus because it give you the confidence that you are actually recording reliably." *196 Maple Avenue, White Plains, NY 10601. (914) 997-1603.*

□**AUDIO TECH RECORDING** (New York City), owned by **Joe Bankowski**, is a 24-track audio-for-video production facility. Equipment includes an **APSI 36/32** console, a **Soundcraft SCM-762** multitrack, an **Otari MTR-12** half-inch, and **Ampex ATR-104** half-inch/four-track, and an **Otari MX-5050B** two-track. Outboard gear comprises an **Eventide H949 Harmonizer**, **Lexicon Super Prime Time** with memory, **Deltalab Acousticomputer DL2**, four **Lexicon PCM-42s**, 10 **Drawmer 201** dual gates, four **Valley People Dynamites**, two **UREI LA4s**, four **dbx Model 165As**, **Aphex Aural Exciters**, and **Aphex Compellor**. A complete MIDI network includes a **Roland MSQ-700**, **SBX-80** SMPTE controller, **Yamaha DX-7** synthesizer, **Oberheim OB-8**, **DSX**, and **DMS**, **Yamaha CXF-5** computer for the **DX-7**, and the **DX-9**. Control room dimensions are 15 by 24 by 13 feet, and the studio measures 40 by 26 by 19 feet. *548 Eighth Avenue, Suite 602, New York, NY. (212) 719-9540.*

□**SHEFFIELD AUDIO/VIDEO** (Phoenix, Maryland) has installed a new **Solid State Logic SL4000E** automated console with **Primary Studio Computer** and **Total Recall** in Studio A. According to audio manager **Richard Van Horn**, "We are using the new console for both recording audio sessions, as well as for Audio-with-Video. We just got a contract for DIR Broadcasting for the *King Biscuit Flower Hour*." Regarding the equipment selection, Van Horn says, "we picked up an SSL board because I got nothing but rave reviews from everybody that has one. That sold me all by itself. Plus, because we like the board." The new SL400E is linked to a **Studer A800** 24-track via a **BTX Softouch** synchronizer. *1318 Sunny Brook Road, Phoenix, MD 21131. (301) 628-7260.*

Southeast:

□ **QL MOBILE RECORDING** (Coral Gables, Florida) is operating "Criteria's Wheels," according to an agreement reached between the QL and Criteria. The mobile recording trruck will be equipped in the near future with a 32-track **Mitsubishi X-800**, as well as standard 24- and 46-track analog configurations. According to QL president and chief engineer **Rob Burr**, who plans to offer digital multitrack rates competitive with current analog remote rates, the 26-foot GMC Trans Mode Van currently is equipped with a digital-ready, customized MCI JH-600 Series 36 in/out console, twin MCI JH-24 multitracks and/or Criteria's digital multitrack. Other onboard gear includes a large collection of microphones, delays and effects. Two 27-input mike splitters and 500 feet of snake will enable the truck to interface easily in any production situation, Burr adds. Now that **Century One Productions**, a full-service

video production firm, has joined forces with Criteria Studios, the facilities available at the new Criteria complex are described as being among the finest anywhere in the world. Burr, who has been using digital technology for the last three years, sees the remote recording market eagerly embracing multitrack digital next year. "Audio-for-Video is suddenly being taken very seriously due to several factors. Stereo Television is a big reason for the rapid upgrading of the audio chain in television broadcasting. Only digital multitrack can deliver that uncompromised fidelity that will soon be the norm. The new Hi-Fi VHS and Beta systems are going to drastically contradict the old adage that the consumer cannot recognize quality audio. Due to their superior signal-to-noise ratio, duplication masters for these mediums *must* consist of high quality video married to digital audio. This means keeping the sound in the digital medium from the beginning to end." *314 Romano Avenue, Coral Gables, FL 33134. (305) 446-2477.*

☐ **DUKE UNIVERSITY RECORDING STUDIO** (Durham, North Carolina) has taken delivery of an **Otari MTR-12C** ¼-inch mastering machine, said to be the first in the region. According to **Frank Konhaus**, studio manager, the 12-inch reel capacity upgrade was in response to classical recording clients' desire for 30 ips, no-NR recording. The studio is equipped with a **Neotek Series II** 20/16 desk, **MCI** multitrack, **dbx, Orban,** and **DeltaLab** outboard gear, and **Dahlquist** monitors. In addition, the facility has added video capability with the construction of a ¾-inch video production room, equipped with a Sharp XC-800 camera, Chyron VP-1 character/graphics generator, **JVC** VCRs and video monitors, and **EECO** timecode editing controller. *036 Biddle Music Building, Duke University, Durham, NC 27708. (919) 684-3460.*

FLORIDA SOUND — new 24-track studio

☐ **FLORIDA SOUND** (Clearwater, Florida) is a new 24-track facility for Tampa Bay area musicians, bands, commercial composers and broadcast advertising producers. Equipment highlights include a **Neve 8068** MkII 32-in/24-out console, **Studer A80VU** 24-track and **A80RC** two-track, **Lexicon** digital reverb, and **UREI** 813B Time Aligns monitors. According to chief engineer, **Gary Rivera,** "Florida Sound is by far the most well-equipped sound recording center on Florida's west coast." *3350 Ulmerton Road, Clearwater, FL 33520. (813) 577-7113.*

☐ **ROAR PRODUCTIONS** (Columbia, Maryland) has purchased a brand new **Toyo** grand piano, which will be the centerpiece of the new wood floor in Studio A. Other additions include two **Symetrix** 522 multipurpose processors, and additional **Sennheiser** 421 and 441 microphones. *6655H Dobbin Road, Columbia, MD 21045. (301) 596-0600.*

☐ **MORRISOUND RECORDING** (Tampa, Florida) has installed an automated 32-channel **Sound Workshop Series 34** board with high-resolution meters, and a new pair of **UREI 813B** monitors. Album projects and jingle productions form the mainstay of the facility's business. *5120 North Florida Avenue, Tampa, FL 33603. (813) 238-0226.*

☐ **SOUTHERN TRACKS RECORDING STUDIO** (Atlanta, Georgia) is a new facility dedicated to the memory of former Lowery Music Group vice president, **Mary Tallent,** who was killed in an automobile accident in December, 1983. Construction of the new studio, located behind the offices of The Lowery Music Group in Atlanta, was necessitated when the former Southern Tracks Studio was torn down to make way for a Metro Atlanta Rapid Transit rail station. Studio design was by **George Augspurger** of **Perception, Inc**. According to Augspurger, the fact that Southern Tracks was built from the ground up enabled him and architect **John Edwards** to include a number of acoustic-enhancing designs and construction techniques in the studio and control room. "In addition to being one of the quietest studios in the country — which is very important in these days of digital recording — Southern Tracks offers five distinct recording areas, each with variable acoustics," he says. The large main room includes three isolation booths, each with variable acoustic panels. The control room houses a **Harrison** automated console with both **Studer** and **Ampex** tape machines. Studio manager is **Mike Clark**. *3051 Clairmont Road, Atlanta, GA 30329. (404) 329-0147.*

South Central:

☐ **DALLAS SOUND LAB** (Irving, Texas) has purchased a **Sony PCM-3324** digital multitrack, and a **Kurzweil 250** digital synthesizer. According to the studio's **Johnny Marshall,** the new 88-note polyphonic synthesizer has the ability to use artificial intelligence to reproduce sampled acoustic sounds. "It is the first digital sampling system with touch-sensitive keyboard to make a more realistic, emulative natural acoustic sound," he states. "A 12-track digital sequencer inside the 250 makes it possible to score demos without ever rolling tape in the studio." The studio also has installed a computer to interface with the Kurzweil 250, which will allow the system to sample any acoustic sound in the studio, spread over the 88-note keyboard, to be used at any time. "Artists who previously could not afford to have large full-scale orchestration because of budget parameters," Marshall adds, "can now afford and can realistically create the sounds of strings and horn instruments." The PCM-3324 was supplied by **Audio Intervisual Design,** Los Angeles. *6305 North O'Conner Blvd., Irving, TX 73039. (214) 869-1122.*

☐ **SOUNDS UNREEL STUDIOS** (Memphis) is a new 24-track facility owned by **Jon Hornyak** and **Don Smith**. Design work for what will eventually be a two-studio complex was handled by **Phase Audio, Inc**., a Memphis-based sound contracting firm. The control room is equipped with a **Sound Workshop Series 30** 28/24 console, **Otari MTR-90 II** 24-track with Autolocator, **Otari MTR-10** two-track, **Ursa Major 8X32** digital reverb, and other outboard gear by **dbx, Lexicon, Roland, Korg** and **Symetrix**. The control room monitors are a custom **JBL/TAD** system, designed and installed by **Steve Durr** of Nashville. *1902 Nelson Avenue, Memphis, TN 38114. (901) 278-8346.*

NASHVILLE NETWORK — Harrison TV-3

☐ **THE NASHVILLE NETWORK** (Nashville) has taken delivery of a second **Harrison TV-3** console for stereo teleproduction and post-production at the country-music cable television network. "The Nashville Network was delighted to find a top-of-the-line, professional audio console that provides the facilities to easily and effectively handle Stereo TV sound," comments **Hugh Hickerson,** director of engineering for Opryland, USA Broadcast Properties, of which TNN is a part. "We wanted to buy quality. And, the Harrison TV-3 provided us the greatest ease of attaining our needs for TV of any non-custom consoles." *2806 Opry Lane Drive, Nashville, TN 37214. (615) 889-6840.*

☐ **DOVE & NOTE RECORDING COMPANY** (Houston, Texas) recently completed a recording session with the **Boston**

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Symphony Orchestra, using **Crown PZM** extensively. D&N supervised the overall project, did the recording engineering and performed the live stereo mix. **Digital Entertainment Corporation** supplied a **Mitsubishi** 32-track **X-800**, while **Digital Services**, Houston, provided the remote studio facilities. *Houston, Texas.*

OMEGA AUDIO — refurbished Heider truck

☐ **OMEGA AUDIO** (Dallas, Texas) has acquired the former **Wally Heider Mobile Unit #2** from The Record Plant, Los Angeles, Omega has subsequently refurbished the facility, based on a 1978 GMC Loadstar, to now include an **Automated Processes**, 38-in/24-out console with 550 EQ and program busses equipped with eight stereo VCA groups; a 600-foot snake with 54 transformer isolated splits; 95 various microphones with models from **Neumann, AKG, EV, Shure, Crown, Sony,** and **Sennheiser; JBL 4430** monitors, with **UREI 539** EQ, and **Auratone** monitors; two **Otari MTR-90** 24-tracks; two **Otari MTR-10-4** two/four-tracks; two **Technics** cassette decks; a **dbx Model 165** limiter, two **model 162** stereo limiters, and two **model 160** limiters; two **UREI 1176LN** limiters; a **Teletronics LA2A** limiter; an **Orban** stereo parametric EQ; an **RTS** two-channel IFB and intercom system; **Lexicon 224X** digital reverb; two **MXR .01a** digital reverbs; and two **DeltaLab Super Time Line** digital delay lines. In addition, the studio has a **Mitsubishi X-80** digital two-track, and will take delivery of an **X-800** early next year. *8036 Aviation Place, Dallas, TX 75235. (214) 350-9066.*

Midwest:

☐ **TONE ZONE STUDIOS** (Chicago, Illinois) is a new recording complex owned and operated by **Jesus People USA-FGM**. The main studio, a 700-square foot room with a 20-foot ceiling, was designed and built as a "rock n'roll room," and has a large marble floor, three isolation booths, and **UREI 813** playback monitors. The 320 square-foot control room is a LEDE room (currently in the process of registration), designed by **Glenn Meeks** from **Comcast Communications**, Indianapolis. Equipment includes and **Otari MTR-90 Series II** 24-track, **MTR-12** half- and ¼-inch two-track and a **Harrison MR-4** 36-input console. Outboard gear includes **dbx, UREI, Symetrix, Drawmer** and **OmniCraft** compressers/limiters/gates; **Orban** and **Ashley** equalizers; **Lexicon, Eventide** and **DeltaLab** delays. A **Studio Technologies Ecoplatl 1** and **Quantec Room Simulator** are used for reverb. Monitoring is handled by a pair of **UREI 6500** power amps and **UREI 813-B** Time-aligned speakers; **Yamaha NS-10s** and **Auratones** are also available for auxiliary monitoring. Studio playback and cue mixes are powered by **Hafler** amps. A full compliment of **Neumann, AKG, Sennheiser, Crown, EV** and **Shure** microphones are available, as well as some vintage tube microphones. Available instruments include a **Yamaha C-7 Grand piano, Yamaha DX-7** and **Roland** synthesizers. **Simmons** and **Linndrum** machines with trigger interface. The engineering and maintenance staff is currently comprised of **Roger Heiss** and **Roy Montroy**. *4707 N. Malden, Chicago, IL 60640. (312) 664-5353.*

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☐ **UNIVERSAL RECORDING** (Chicago) has taken delivery of a **Mitsubishi X-80** digital two-track. According to Universal president, **Murray Allen**, the machine was chosen for its simplicity of operation and advanced editing capabilities —which he sees as being superior to those currently found on other digital machines. "The engineers here really enjoy working with it because it's similar in many ways to an analog recorder," he says. "They can splice and edit the tape without any problems." The facility has already used the X-80 for various mixdown projects, including several TV commercials for McDonald's and an upcoming album by Peter Brown. **Bill Bradley**, who engineered the digital recording of Brown's LP, was impressed with the way the X-80 handled the mixdown from the two 24-track analog recorders used in the project. He particularly liked the X-80's direct off-tape monitoring ability. "I did most of my mixing through the machine," he says, "and, when I listened to it back off the console, the differences were noticeably slight. From a tape handling standpoint, it's much quicker and easier to use than other digital machines I've worked with. I can also edit material on the X-80 almost as quickly as I can edit analog — and the end results are always sterling." Allen notes that requests from commercial clients for digital audio masters are steadily increasing, and says that he expects digital audio to have its biggest impact on the emerging Audio-for-Video industry. "We do a lot of work for MTV and other stereo cable networks," he explains. "And, wherever it's conceivably possible, we use digital masters. I think you're going to see digital technology playing a major role in the future of television and motion picture audio." *46 E. Walton, Chicago, IL 60611. (312) 642-6465.*

☐ **STUDIO A** (Dearborn Heights, Michigan) recently acquired the following equipment: **Sony TC-K555** cassette decks, **Lexicon 200** delay processor, **Lexicon PCM-42** delay line, and the sampling system with terminal support for **New England Digital Synclavier II**. In addition, construction has begun on a new 24-track facility designed by **John Storyk Associates**. *5629 N. Beech Daly, Dearborn Heights, MI 48127. (313) 561-7489.*

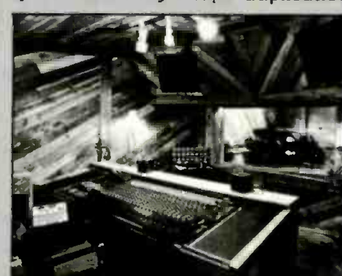
☐ **3001 RECORDING STUDIO** (Columbus, Ohio) has been purchased by **Jeff Gastineau**, owner of **Apple Recording Studio**, Dayton. Apple will move to the 3001 location, and a second studio will be opened with 46-track capability, along with video lock-up. 3001 will offer what is described as one of the most versatile synthesizer systems in the Midwest, consisting of more than a dozen MIDI synthesizers, and computerized sequencing with over 30,000 notes of storage capacity. Gastineau, synthesist with the group Money, says, "3001 will continue to function as a studio where a musician can create music without complications." *3001 Indianola Avenue, Columbus, OH 43202. (614) 262-3001.*

☐ **ARS RECORDING STUDIO** (Alsip, Illinois) has completed construction on its LEDE-type control room, a project that ended with the installation of **Quadratic Residue Diffusors** along the rear wall. The new room was designed by **Doug Jones** of **Electro-Acoustic Systems**, in conjunction with **Gary Cobb** of **ARS Enterprises**. New additions also include a **Bluthner** grand piano. *11628 So. Pulaski, Alsip, IL 60658. (312) 371-8424.*

☐ **CREATIVE AUDIO** (Urbana, Illinois) has acquired the first complete **PPG System** — Wave 2.3, Waveterm, PRK and EVU —in the Midwest. Additional studio purchases include two **AKG C-12s**, a **Neuman U-67**, **Yamaha DX-7**, and **Garfield Electronics Dr. Click**. Independent engineer **Jonathan Pines** reports purchasing 10 **API** equalizers (560s, 550As, and 554s), two **Pultec EQH-2s**, **Helios** stereo EQ, two **Neve 1064** preamp/equalizer, a **Marshall Time Modulator**, Compressor/Limiters by **Trident**, **RCA** (tube) **UREI** (tube), and a mixed assortment of **AKG** and **Neuman** tube microphones. *705 Western Avenue, Urbana, IL 61801. (217) 367-3530.*

Mountain:

☐ **BONNEVILLE MEDIA COMMUNICATIONS** has equipped a second audio control room for video sweetening. The newly remodeled Studio A is fitted with an **Audio Kinetics Q.Lock 3.10-3** synchronizer, and **Ampex MM-1100** 16-track, and an **MCI JH-110** C-Format audio layback machine. Studio manager **Dave Michelsen** says the concept of low-cost audio sweetening made possible by the use of the audio layback machine, which eliminates the need for an expensive video editing bay, has caught on among local video producers. "In fact," he adds, "demand for film looping, spot sweetening and teleconference translation has been keeping our Studio B so busy for the past year, we set up Studio A to handle the overflow." Also, an additional shift, more efficient use of equipment, and several technical improvements have combined to push the capacity of the facility's tape duplication department to 40,000 cassettes a day. Duplication slave units are now equipped with **Dolby HX Pro** circuitry; a **Studer A-80** master recorder, also fitted with HX, has been added to the mastering room. *130 Social Hall Avenue, Salt Lake City, UT 84111-1580. (801) 237-2600.*



ROSEWOOD — new digital synthesizers

☐ **ROSEWOOD RECORDING COMPANY** (Provo, Utah) has placed a new emphasis on digital keyboards with the addition of an **E-mu Systems Emulator II** sampling synthesizer and **Yamaha DX-7** with **Apple Computer Interface**, according to owners **Guy** and **Kristen Randle**. The Emulator II's memory permits the user to sample a real instrument at multiple points along the keyboard so that the sounds may be reproduced faithfully in terms of timber and modulation. The keyboard may be programmed for touch sensitivity in both amplitude and attack. The DX-7 Apple interface makes for easy preset selection and programming on the computer screen, and the initial software provides over 900 sounds, catalogued by instrument type and performance group. With MIDI interface, the player has a multitude of real and realistic sounds literally at his fingertips. Also recently acquired were an **E-mu Systems Drumulator**, **Yamaha NS-10** monitors, **Eventide 910 Harmonizer** and **Lexicon 200** digital reverberator. *2288 West 300 North, Provo, UT 84601. (801) 375-5764.*

Southern California:

☐ **GLEN GLENN SOUND** (Hollywood) has become the first independent post-production house to acquire a digital multitrack recorder. "The purchase of the **Sony PCM-3324** multitrack is a natural extension of our relationship with Sony," explained **Rick Larson**, Glenn's senior VP. "We feel that Sony has established itself as the leader in digital technology, and they have shown the necessary interest in translating the technology to the film business." The sale was handled by **Audio Intervisual Design**. According to **Tom Kobayashi**, president of Glen Glenn, "The PCM-3324 is used as any other recorder would be for dubbing, editing and mixing; we didn't have to change our normal operation. Now we have the advantage of no generation loss when re-recording, and also a tremendous leap forward in the clarity and impact of the finished soundtrack. It's clear to us that the future

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of sound for film and TV will be digital; we just happen to be leading the way." *909 North Seward, Hollywood, CA 90038. (213) 467-7221.*

☐ **LION SHARE RECORDING STUDIOS** (Los Angeles) has added an **AMX DMX-1580S** pitchshifter and **RMX-16** digital reverb to its list of outboards. *8255 Beverly Blvd., Los Angeles, CA 90048. (213) 658-5990.*

☐ **EFX SYSTEMS** (Burbank) has appointed **Jere Mendelsohn** to the position of studio manager. A graduate of Northwestern University of Chicago, where he received a B.A. in Radio, Television and Film, Mendelsohn then moved to California and graduated from the Guitar Institute of Technology in Hollywood. He will be responsible for the studio's day to day operation. *919 N. Victory Blvd., Burbank, CA 91502. (818) 843-4762.*

☐ **LARRABEE SOUND** (Los Angeles) has added an **AMS DMX-1580S** pitch shifter with 4.8 seconds of delay, keyboard interface and chorus controller, and an **RMX-16** digital outboard. In the last six months, other equipment additions have included a **Mitsubishi X80** digital two-track, two **Studer A800** multitracks to complement the facility's **Solid State Logic** computerized console, and **Audio Kinetics Q.Lock** synchronizer. *8811 Santa Monica Blvd., Los Angeles, CA 90069. (213) 657-6750.*

☐ **CALIFORNIA RECORDING STUDIOS** (Hollywood) recently upgraded Audio/Video sweetening facility with the installation of a **BTX Softouch** SMPTE synchronizer. The studio recently handled post production for the underscoring and source music for CBS Television's *Dreams*, engineered by **Tim Garrity**. *5203 Sunset Blvd., Hollywood, CA 90027. (213) 666-1244.*

Northern California:

☐ **MARS RECORDING STUDIO** (Aptos), now operating in its 10th year of business, has updated its selection of outboard gear with a **Lexicon 200, Studio Technologies Ecoplate II, Eventide Harmonizer** and a **Lexicon Prime Time**. The studio has also complemented its 16-track facility with new **UREI** Time Align speakers and **Neumann U87** and **U67** microphones. The desk now boasts new **Super Audio Amps** and eight echo returns. *5944 Freedom Blvd., Aptos, CA 95003. (408) 688-8435.*


MUSIC ANNEX — new Otari tape duplicator

☐ **MUSIC ANNEX** (Menlo Park) reportedly has become one of Northern California's largest tape duplicators with the recent acquisition of a new **Otari DP-80** high-speed 64:1 system. A total of 4,000 audio cassettes per day can be produced with the current system, which includes one Master Reproducer accepting 7.5 ips masters running at 480 ips, and two slave recorders. The facility has also recently ordered two more additional slaves, which will bring their daily capacity up to nearly 8,000. "Quality is very high," states owner **David Porter**, "because the DP-80's 7.5 ips master speed can maintain a high-frequency response while yielding a 64:1 production ratio. We expect to made over 750,000 high-speed and real-time cassettes in 1984." *970 O'Brien Drive, Menlo Park, CA 94025. (415) 328-8338.*

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NASHVILLE IN THE EIGHTIES: BACKWATER OR BELLWEATHER?

A Look at the Changing Recording and Production Scene in Music City USA

by Sam Borgerson

Without a doubt, many engineers and producers working within the recording industry consider Nashville to be the future recording center for a wide range of music. Talented people are coming from all over the world — London, New York, and in particular, Los Angeles — to become part of the action. If you are looking for digital 32-track and a fully-automated console hooked up to a room full of digital synthesizers, then the hills of Middle Tennessee are the place to be. And, yes, digital *is* running rampant across the country music charts.

However, some less charitable types will tell you that, as a recording center, Nashville is hopelessly mired in the old ways. Compared to New York or Los Angeles, for example, Nashville is often considered to be something of a backwater where studios are ill-equipped and poorly maintained. "Good Enough for Country" is the everyday excuse and one, they might argue, that justifies mass production of a homogenized "Nashville Sound" though the use of well-worn and unimaginative recording techniques.

So who is right? What is really going on here? What is the "Nashville Sound" of the Eighties, and how are the engineers and producers of Music City USA making it?

Before approaching these questions, it might be helpful to take a whirlwind tour through the history of recording in Nashville during the past 20 years. Nashville, after all, is a place where tradition is just as important as technology — and sometimes where tradition is at odds with the prevailing technology.

A Measure of History

Throughout the decade of the Sixties, the Nashville recording scene was dominated by the label-owned studios, principally RCA and Columbia/CBS. Under the guidance of producers like Chet Atkins and Billy Sherrill, these two studios were responsible for turning out an astounding number of country hits. Pop recording also kept the reels turning in these rooms: the lion's share of hits by Elvis Presley, Roy Orbison, and the Everly Brothers were cut at RCA, while in the late Sixties Bob Dylan and the Byrds checked in at Columbia.

Technologically, Nashville's label studios were not significantly different from their sister studios on the East and West coasts: they used the same custom consoles, often made in-house; they used the same three-, then four-, then eight-track tape machines; and, of course, their engineers belonged to the same union.

By the end of the Sixties, the baby-boomers had the whole music business booming like crazy. Good-sized budgets for pop and country sessions were spilling over the booked-up label studios; independent studios like Quadrafonic, Woodland, and Jack Clements jumped into the major-label recording game.

About this time Nashville accepted its First Wave of Immigrants, most of whom migrated from neighboring cities in the South. Pickers-turned-producers Norbert Putnam and David Briggs came up from Muscle Shoals, Alabama, while pop/soul hitmaker Chips Moman came over from Memphis to erect a second American Studio on Music Row. Their experience in pop and R&B production helped sustain a flow of pop artists (and their larger budgets) through the Nashville studios in the late Sixties and early Seventies.

In the mid-Seventies, thanks to multitrack product from MCI's Jeep Harned and David Harrison of Harrison Systems, everybody got into the act. What the heck, if David Briggs and Norbert Putnam can make hit records, such as Neil Young's *Harvest* and Dan Fogelburg's entire first album in an old house renamed Quadrafonic, and if an MCI 16-track can be had on a beer budget, anybody can do it! The local studio suppliers did land-office business, putting package studios into every nook and cranny up and down Music Row. Sometimes the studio entrepreneurs demanded a well-designed, technically first-rate facility. Alas, sometimes they didn't.

But studios across the total quality spectrum stayed busy under the motto, "Hey, it's good enough for country!"

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NASHVILLE IN THE EIGHTIES

For four or five years, everybody was happy, except for the accountants at RCA — so many of the label's artists were going elsewhere that its own studio had become uneconomical. In 1977 RCA bowed out, selling the studio to Owen Bradley, who turned it into the Music City Music Hall.

Then, as Nashville rolled over into the Eighties, the studio community was hit by two major setbacks. First, taken as a whole, Nashville studios had fallen far behind New York and L.A. in adopting new technologies and recording techniques. The huge recording budgets of the mid-Seventies had largely bypassed Nashville; there was no incentive to build million-dollar-plus facilities, because nobody could afford to pay the hourly session rates. So none were built. And, second, the prevailing economy and the advent of video games had sent the whole record business into a tailspin. In 1982, CBS decided to turn the legendary Columbia Studio into offices — the label studio era was over.

But behold, a Second Wave of Immigrants had come from Los Angeles to lead Nashville into the promised land of big budgets and new technology. First and foremost was label executive (Elektra/Asylum, then Warner Brothers, now MCA) and producer Jimmy Bowen, followed by fellow producers Rodney Crowell and Jim Ed Norman. Trailing close behind them was a phalanx of L.A. engineers, including Joe Bogan, Donivan Cowert, Eric Prestidge, and Dave Hassinger. Under Bowen's crusading leadership, the L.A. refugees have helped push Nashville back into a close technological race with the best studios on the two coasts.

So, was this influx of new blood responsible for Nashville's rebirth? No, that came in with the Third Wave of Immigrants . . . from Belgium.

However, before we head out to the hills to visit the Belgians in their digital castle, let's consider the current state of recording techniques and studio technology in Nashville.

The City Comes of Age

Laying down basic tracks in Nashville studios involves numerous variations on two basic approaches: the quick, low-budget "country" approach; and the more painstaking "pop" approach. The quick approach dictates the use of a drum booth and considerable isolation of instruments. Microphone set-up in the booth is modified only slightly from session to session, and achieving the right drum sound often takes less than an hour. The "pop" approach, on the other hand, involves placing the drums in a large, live room, and tailoring the mike placement to the sound of the room and the style of the drummer. As budgets permit, the latter approach is becoming increasingly popular.

Veteran engineer **Ernie Winfree** has watched the changes over 15 years of Nashville sessions. "When I started here the most important thing was isolation," he recalls, "but the overall trend in Nashville seems to be going away from that isolated sound. Leakage in itself is not bad, if it's controlled. In some cases it enhances the overall sound."

A relative newcomer on the Nashville scene, engineer **Rick McCollister** is known for favoring a very open and natural drum sound, particularly on his work with Gail Davies. Given the chance, McCollister prefers to use numerous mikes to spotlight each part of the drum kit, but he has also developed a minimal mike alternative for sessions when time is a factor. "You can do a lot with a maximum mike technique," he considers, "if you do it right. But it sounds horrible if you don't have the time to make it work. So sometimes I'll use just five or six mikes pulled back away from the drums to get a natural spread. This way I can get a good, basic sound much more quickly."

Another basic change in Nashville recording technique has been the increased use of overdubbing. **Lou Bradley** witnessed the change during his 13-year tenure at the CBS studio: "Studio B, the old Quonset hut, was a very good room for cutting live country sessions. But now everybody does more overdubbing [of] things they used to do live on the date — harmonica, steel, fiddle, and other fill instruments. That's the growing influence of the California style on Nashville. I personally think there's a spirit, a feeling among the players that is lost by doing it that way. It can be done with overdubs, but I think it's harder to get."

Mixing techniques on Music Row have split the recording community into factions: a growing minority who use automation; and — depending on your point of view — a stubborn or "golden-eared" majority who eschew the computer. "I don't think the automation has been used more than six times since we got it," reports **Paul Goldberg** of the Music Mill. "Sometimes it's because of the track usage, and sometimes because we don't have the time to set it up."

"I personally prefer to use the automation," admits Ernie Winfree, who does most of his work on the MCI JH-500 boards at Soundshop, which are equipped with JH-50 automation and Valley People VCAs. "But I use the [non-VCA] straight-wound inputs

Rick McCollister *(left)* **in his Nashville-based digital/analog basement workshop and personal-use studio. Ernie Winfree** *(right)* **at Sound Recording Studio's automated MCI JH-528 console.**

Photo by: Sam Borgerson

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LEDE DESIGN FOR ACORN SOUND RECORDERS
Incorporating Reflection Phase Grating Diffusors on the Rear Wall to Enhance Spatial Imaging and Stereo Ambiance

by Bob Todrank, president, Valley Audio, Nashville

Editors Note: The author was asked to briefly outline his use of the RPG diffusors at Acorn Sound Recorders, and the subjective listening results from their use. This piece is not intended, however, to represent a definitive article on LEDE™ or RPG™ diffusors.

I would caution the casual reader against believing that all rooms with a "dead" front end and a "live" back end meet the LEDE design criteria. Many years of work by dedicated acousticians, theoreticians, mathematicians, designers and interested educators have gone into the refinement of the LEDE design concepts. Strict criteria have been established governing the certification of a control room as LEDE; we must all avoid insulting the years of dedicated work by a group of fine professionals by labeling a control room as LEDE when, in fact, it is not.

Application of Reflection Phase Grating Diffusors

The use of the RPG diffusor system at Acorn Sound Recorders resulted from a chance meeting between Peter D'Antonio and myself at the New York AES Convention in the Fall of 1983. Having been introduced at a small gathering of people interested in LEDE control room design, we found ourselves later that evening discussing acoustics over dinner. In talking with Peter about my upcoming project for The Oak Ridge Boys, he suggested that I consider his diffusion ideas for the rear wall. We had previously considered some customized form of polycylindrical diffusion for that area. Peter had, up to that time, only built a rough model of his new diffusor system for his own personal studio, and was interested in trying the idea in a larger LEDE environment. Needless to say, I was more than happy to oblige. Thus, Acorn Sound Recorders has the honor of being the first commercial recording studio in the world to install RPG diffusors.

... continued overleaf —

Rear-wall RPG diffusers installed at Acorn Sound Recorders are built from half-inch plywood and 0.040-inch aluminum; the face of each section is covered with plexiglass mirror for a "high-tech" appearance. The lower (horizontal) section is built from four, two-by four-foot units to form a complete surface that is four by eight feet by 18 inches deep. The top (vertical) section also is made up of four units for a total area measuring four by eight feet. The side mounted plexiglass panels — designed by Valley Audio, and referred to as "Hass Kickers" — are intended to present sharp, specular reflections into the mixing position, and are set up for a specific delay time. Their operation compliments the later arrival of the diffuse reflection patterns derived from the RPG panels.

NASHVILLE IN THE EIGHTIES

to go to the recorder when cutting tracks, and the faders just for monitoring. That way I eliminate one generation of the VCA sound."

Over at Emerald Sound, chief engineer and L.A. expatriate **Joe Bogan** refuses to compromise. "I never found a VCA console I really liked," he insists. "I think it colors the sound. Also, automation takes some of the personality out of the mixing."

Bogan's view seems to be shared by a clear majority of Nashville mixers, and their opinions have prompted many studios to invest in non-automated Neve and Trident consoles — new and used — rather than automated consoles in the same price range.

Nashville has gone through monitoring trends at a disturbing pace. Certain brands have been in then out of favor; Meyer Sound cabinets, in particular, now appear to be the vogue. But, most of all, perhaps in utter despair, mixing engineers seem to be turning to small, "close-field" bookshelf speakers mounted on the console's meter bridge, as exemplified by the ubiquitous Auratone Sound Cubes.

"There is a trend here away from the horn-oriented systems," reports McCollister, "and more toward phase linearity in the systems. My personal favorite is the system at Master Mix, which is a quad-amped, four-way system designed by Claude Fortier and installed by Neal Muncie. It's an all-cone system, and will play *very* loud. For my own basement workshop/-studio though, I have John Meyer 833 cabinets, which are very phase accurate. Also, I think there's a trend in the newer rooms to eliminate room equalization as much as possible."

If there is a monitoring standard in Nashville right now, it would have to be the Yamaha NS-10. These bookshelf speakers are inexpensive enough that everybody can have a pair, and most engineers offer that it is hard to get way off base with them, just so long as you remember that the top-end response becomes brighter if they are pushed hard.

Some people, however, like to hedge their bets. On the liner notes of the latest release from Ronnie Milsap — Nashville's acknowledged sound and equipment connoisseur — you see Ronnie and his engineers hunched behind stacks of console-top speakers, including Auratones, Yamahas and Electro-Voice Sentry 100s. "We were going through our paranoid period I guess," chuckles **Ben Harris**, chief engineer at Milsap's Groundstar Laboratory. "We used the E-Vs most of the


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LEDE DESIGN FOR ACORN SOUND RECORDERS — continued . . .

For those unfamiliar with the terms, LEDE (Live-End/Dead-End) is an acoustical design concept with specific applications to audio recording control rooms. The LEDE approach primarily consists of a Reflection Free Zone in the front of the control room to minimize acoustic phase and time "distortions," while providing a "live" and very diffuse sound field in the rear of the room. RPG Diffusors provide a significant improvement in wide-angle acoustical dispersion over a broad band of frequencies. Since detailed technical descriptions covering these subjects are readily available, we will limit this short discussion to the subjective results of using the RPG diffusors on the rear wall at Acorn.

Webster defines the verb diffuse, "To pour or send out so as to spread in all directions"; and diffusion as to "A scattering, dissemination, or dispersion". According to Peter D'Antonio, "The live end [of the control room] is achieved by positioning RPG diffusors on the rear wall in such a manner as to reintroduce the energy passing the mix position, after an initial time delay, temporally and spatially diffused." In other words, part of our objective in the LEDE concept is to return a significant amount of acoustic energy (sound) from the rear wall back into the mix position. This energy must be widely scattered over a broad frequency spectrum, so as to not introduce frequency response anomalies. If the energy can be returned into the mix position within the proper time window, and is sufficiently diffuse, we can greatly enhance the room "ambiance" and spatial imaging.

Practical Improvements in Sound Quality

As to the practical side of all this, I was extremely fortunate to have evaluated our work at Acorn before the RPG diffusors were installed in the rear of the room. Our first listening tests were as expected: Good left/right balance; smooth frequency response; minimal phase anomalies; and a solid center image. After the proper installation of the RPG diffusors on the rear wall we were astonished by the audible improvements. First of all we improved the overall room "ambiance." We could clearly hear more "liveness without coloration," which led to a more pleasant musical experience. We greatly improved the listening environment over the entire back half of the control room. No longer was the mix position the only worthwhile listening spot. Listeners could comfortably move about the entire back half of the room, and maintain a reasonable representation of tonal balance and stereo image.

Another advantage was the dramatically improved stereo image across the entire console face. While sitting in the producer's chair at the extreme left of the console, I could hear a complete stereo panorama, including a solid right speaker and everything in between. We've all sat behind a left-handed producer's desk, and had the right speaker virtually disappear. At the mix position one could hear a significant improvement in the perception of stereo panning. We could perceive console panpot placement to a much finer degree than normal, which translates into a more reliable "stereo" image. Many listening environments provide us with a good left and right image and maintain a solid center, but few provide us with a strong sense of space and image in between. The image was also perceived to be much wider in space. In other words, the stereo spectrum extended beyond the left and right monitors to give the impression of a wider panorama. This perception seemed to be somewhat dependent on choice of program material.

Frequency response, and therefore tonal balance, was also maintained over a wider range of sound pressure levels in the room. Even at very low levels, the tonal balance remained. Most listeners also remarked that ear fatigue was reduced and that they could monitor at lower SPL levels while maintaining their impression of the program sounding louder.

There was an even more significant observation concerning the area of spatial imagery (the feeling of depth front/back and up/down). The sound truly became three-dimensional, giving the listener a perception of space behind, in front of, on top of, and underneath the stereo panorama. We feel that this increased spatial imagery greatly enhances the overall listening/monitoring experience.

Of course we must realize that much of this perception exists through the proper implementation of the LEDE design concepts, and not all from the RPG diffusors. We felt, however, after many hours of listening tests that the RPG diffusors offered a major contribution to this overall perception. They directly caused some of the aural phenomena outlined above and greatly contributed to the enhancement of others.

So what does all this mean to the music which, of course, is the reason we all participate in these exercises? Fresh ideas, improved listening techniques, and ever increasing musical awareness must become a part of our continuing educational process. With the coming age of better digital recordings, electronic instruments,

Interior of Ronnie Milsap's Groundstar Studios, equipped with a Neve Model 8128 with NECAM 96 console automation, and two Studer A800 multitracks.

time, but we had the others up there for those times when Ronnie would say, 'Oh yeah, well let's see what happens when we feed it through *these*'."

Moving Beyond Country

"Most country records sound very good, compared to when I first came here," says **Gene Eichelberger**, who worked on pop/rock sessions (primarily with producer Norbert Putnam) for most of his years in Nashville. "For a long time you heard mainly the vocal on country records — a very up-front vocal sound. Little attention was paid to the drum or instrumental sounds. Back then, since we were working with pop budgets, we had the time to get those sounds that you can't get with a quick overdub. But that way is catching on; now people here know how to strive for excellence."

There are several speeds and processes for putting nanoWebers on tape, and Nashville's preferences in that regard also have changed recently. For most of his years at CBS, Lou Bradley cut everything at 15 ips with Dolby, a combination he liked. Using 30 ips with the older machines caused low-end problems, he claims, whereas Dolby alignment was not a problem as long as the tapes stayed at CBS — which they usually did.

Today, running 30 ips without noise reduction is the *de facto* standard. Bradley says the newer tape machines have a good LF response at 30 ips, and noise reduction still can pose a problem if tapes move from studio to studio.

Rick McCollister, who just *hates* noise, is about to abandon analog multitrack altogether, but for the time being runs at 30 ips *with* Dolby. "I have to get the noise floor *way* down because I use what most people would consider an abnormally low program level," he explains. "By not loading the tape I can keep my upper dynamics more realistic." McCollister, as we shall see, is ripe for digital technology.

And that's an indication of the way they do things in Nashville these days. The studios wherein they do it are also undergoing some changes, with one eye looking back at room acoustics of yesterday, while the other flirts with the hardware of tomorrow.

Current Design and Equipment Trends

To understand the studio design trends occurring in Nashville today, you have to go back to the Seventies and the Quadrafonic revolution. In the years before Quadrafonic opened its doors, recording in Nashville was done mostly in big, live rooms — RCA, CBS, Woodland, Jack Clements, etc. Then Norbert Putnam and David Briggs took an old house on Grand Avenue, deadened the rooms for maximum isolation, put in a Quad-Eight board and an Ampex 16-track, and proceeded to make Gold and Platinum pop albums with disconcerting regularity.

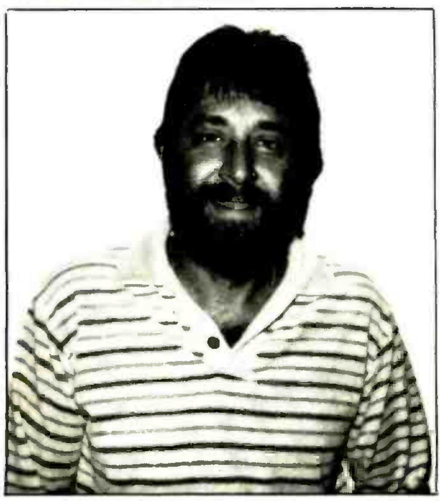
"Quad wasn't a great studio design," admits Gene Eichelberger, who worked there for nearly a decade. "But we made it work. It was a good mixing room because it *always* sounded good in there; you could hear what you were doing. It was an early attempt at a Live-End/Dead-End™ room, come to think of it. It was also very small, so the monitors were practically near-field; you would mix about five feet away from these gigantic monitors.

"Another thing that made Quad great was the way Norbert and David would go for the new equipment as soon as it came out. They'd buy it, raise the rates a little bit, and have it paid for by the time other people found out about it."

By the end of the decade, the isolation concept had fallen out of favor. Putnam left Quad in favor of the larger and more live-sounding Bennett House. Large studios that had deadened their rooms went back to liven them up. And two of Nashville's newest rooms, Treasure Isle and Emerald, were designed as live "rock 'n' roll" rooms from the outset — with Emerald patterned more or less on current L.A. trends, while Treasure Isle leaned more toward the English concepts. [See *R-e/p* October 1983 issue for a full rundown of the design and construction of Treasure Isle —*Editor.*]

Control rooms have been shifting toward the Live-End/Dead-End design philosophy, with larger control rooms also coming into favor. Treasure Isle, Master Mix, and the Oak Ridge Boys' Acorn Studio are three new LEDE-type control rooms in town. [Of the three environments, so far only Acorn has been fully certified as an LEDE design by Synergetic

Engineer Gene Eichelberger has a wide experience of Nashville sessions, including formative years at Norbert Putnam and David Briggs' legendary Quadrafonic Studio built in the Seventies.



Editorial Note: Live-End/Dead-End™ is a trademark of Synergetic Audio Concepts, the organization responsible for offering full certification to control rooms that have been designed according to strict LEDE principles.

LEDE DESIGN FOR ACORN SOUND RECORDERS — continued . . .

superior home playback systems, and new forms of music; we must be better prepared to "hear" what we are recording, especially in relation to "time" information. This is an area of music we are just now beginning to measure. Just like the performance of a symphony orchestra in a great hall, "time" cues in recorded material give us the perception of musical size, space, and depth. The preservation of these cues is vital to the overall enjoyment of any musical performance. Thus the control room of the future must provide its occupants with this increased spatial awareness. I trust all of them will.

RPG™ is a registered trademark of RPG Diffusor Systems, Inc. LEDE™ is a registered trademark of Synergetic Audio Concepts. A bibliography of reading materials covering LEDE design concepts may be obtained from Bob Todrank, Valley Audio. P.O. Box 40743, Nashville, TN 37204. Additional technical information on RPG diffusors may be obtained from RPG Diffusor Systems, Inc. 12003 Wimbleton Street, Largo, MD 20772.

Photo by: Beth Gwinn

Treasure Isle's control room features a 32-input Trident Series 80 (with additional A-Range input modules), Studer A80 multitracks and half-inch mastering machines, plus a large collection of outboard delay lines, and reverb.

Audio Concepts — *Editor*.] Acorn Studio is the first control room in the country to employ a new rear-wall diffusor system designed by **Peter D'Antonio**. According to Valley Audio's **Bob Todrank**, who designed the room, the diffusor creates optimum sound dispersion around the console area. [See accompanying sidebar for further details concerning Acorn Studios' diffusor system design — *Editor*]

In Nashville's first-line master recording studios (as opposed to demo studios), the equipment trends have been decidedly European. Studer moved its U.S. headquarters to Nashville in 1974, and the initial reaction to the expensive Swiss machines was just what you'd expect: "You don't need that for country." Slowly but surely, however, Studer has moved to virtual dominance of the market, with roughly 80% of the major Nashville studios now all-Studer equipped.

The console trend, again among first-line studios, has been away from domestic manufacturers and toward English brands — Neve, Trident and, very recently, Solid State Logic.

Woodland Sound Studios established the Neve tradition in Nashville, with Emerald and Sound Stage following later. Shortly after opening his Groundstar Laboratory, Milsap upgraded to a Neve 8078, which now has left for a new home at A&R in New York to make room for a new 8128. The new board, one of the first to be equipped with NECAM 96 automation, reportedly was ordered because it has sufficient inputs to accommodate 48-track work, rather than for sonic improvements.

Trident has also done well of late; the big-money studios are going for the TSM Series, while operations working on a limited budget have opted for the Series 80. The Bennett House boasts one of the legendary Trident A-Range boards, while Treasure Isle has unearthed some A-Range input modules to supplement its new Series 80.

Solid State Logic also is beginning to penetrate the Nashville market. The first SSL board, a 4000E equipped with Total Recall, was not installed until Bullet Recording took the plunge 1981, and the city's second SSL, also a 4000E with Total Recall, went to The Castle earlier this year.

Nashville's struggle to catch up with New York and L.A. in the studio equipment race is, and always has been, tied to budgets for country records. Although a recent survey puts the average recording budget at over $35,000 per album project, the city attracts very few of the $100,000-plus projects that are needed to support the "super studios."

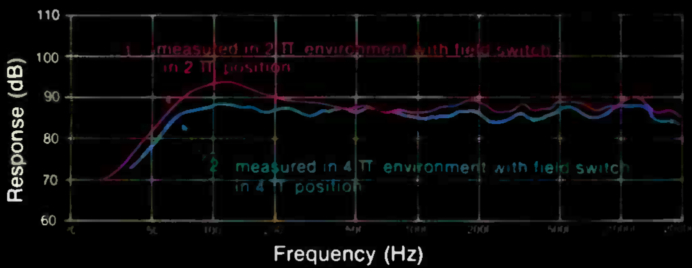
Jimmy Bowen is out to change all that, however. "When I first came here our budgets were $18,000 to $25,000 to make an album," the producer recalls. "On my first Mel Tillis album I spent $36,000 and he stuttered for an hour. 'But, B-B-Bowen, th-that's t-t-twice as much!' His previous album had sold 40,000, and the one I did sold 140,000.

"Well, some people around here still think I'm a damn fool. They say I

FOSTEX FOUR-PAGE INSERT FOLLOWS ON PAGES 127 to 130

FOSTEX RP TECHNOLOGY

FOSTEX



Monitor response will always vary from a 2 pi steradian environment (wall or soffit mounting) to a 4 pi steradian envirorment (free field or console mounting). The Fostex RM-Series Point Source Monitors provide a front panel Field Switch to provide the same response in both environments.

The newest application of the Fostex RP Technology is the tweeter design of the RM-Series Point Source Monitors. Unlike conventional dome or cone tweeters, the flat planer design of the RP tweeter, combined with its unique, low-mass suspension system, permits the entire surface of the diaphragm to be driven with absolute uniformity.

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Another practical benefit of the RP design is the unusually flat impedence curve which results. These tweeters are far less susceptable to RF burn caused by clipped amplifiers and fast winding tape.

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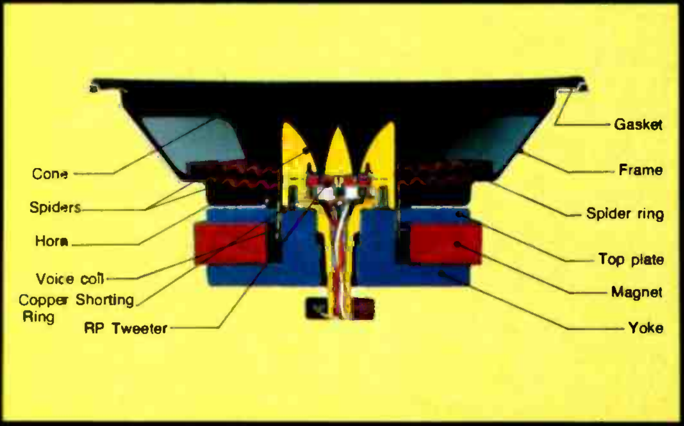

As stereo TV becomes a reality, point source monitoring will greatly enhance imaging in post-production — not only in terms of traditional left-to-right positioning, but also in terms of front-to-back location.

Fostex near field monitors are much more than a playback reference. EQ adjustments, especially at higher frequencies, are truer and more predictable because there is less distortion and there are far fewer phase irregularities than with typical console monitors.


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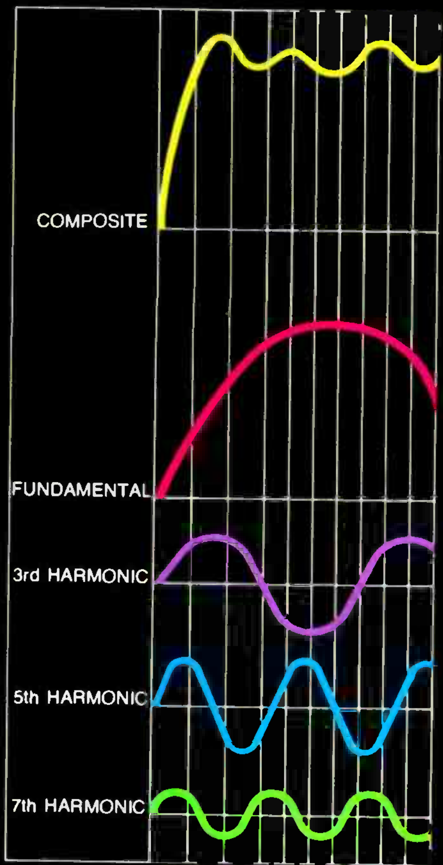

The completed figure to the left is a composite illustration of how a transient pulse looks when the fundamental, 3rd, 5th, and 7th harmonics are all in phase.

The completed figure to the right is a composite illustration of how that same transient pulse looks when the 3rd, 5th, and 7th harmonics are slightly out of phase with the fundamental.

In a typical monitor, the transient sound is smeared because the harmonics arrive ahead of the fundamental. The coaxial design of the RM-Series monitors delivers all the music with complete phase integrity.

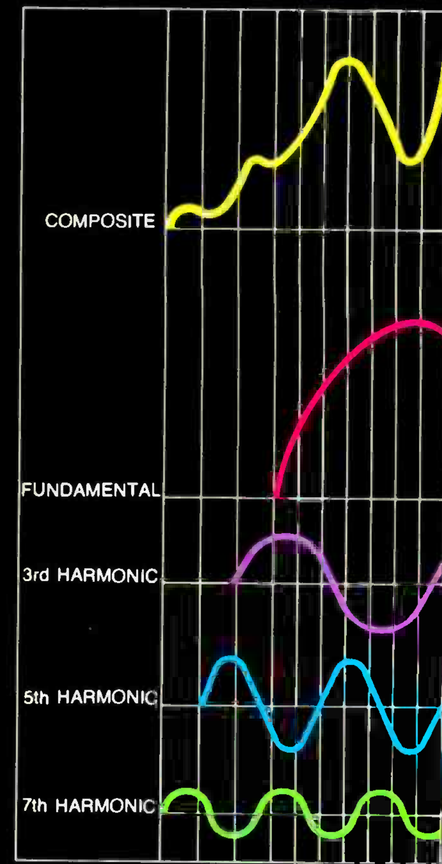

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Photo by: Beth Gwinn

The Castle Recording Studio, located in a secluded 35-acre country estate, features a 48-input Solid State Logic SL4000E console, linked to 3M M81 digital 32-track and four-track tape machines, plus a Studer A80 MkII analog multitrack. Synchronization is handled by a Studer TLS2000 system. A separate pre-post-production and synthesizer facility offers a Fairlight CMI, Yamaha DX-1 and DX-7, Roland Jupiter 8, Oberheim DMX, and Simmons electronic drums. Pictured left at the SSL console are (L to R): engineer Bob Bullock, studio manager Joseph Nuyens, and producer Eddie Kilroy.

came here and taught them how to make a $30,000 album for $130,000, but when Hank Williams Jr. and I got together he wasn't selling records, and now he's selling millions. So maybe I'm *not* crazy after all!

"Studios in Nashville just have not been getting the benefit of the new recording technology because of these small budgets," he continues. "It was all the studios could do just to pay their people and keep the gear running; they couldn't go out and buy a new $5,000 or $20,000 'toy.' So I think that, in order to survive, they started saying, 'Oh, you don't need that for Country.' The one statement I hated *most* was: 'You don't need that for Country.'

"When I came here there were still a lot of people on 16-track when I wanted 24-track. 'Oh, you don't need that for Country.' I wanted digital reverb, and they said 'You don't need that for Country.'

"I soon became belligerent on the other side. I *knew* you couldn't hurt country by making it sound better, by using the innovations that come along. I started using JVC digital on all my mixes 3½ years ago, and I heard the same thing! 'That damn Bowen, he's just wasting money!'

"Remember, they had been used to spending three days and $25,000 here to get a record. But you *have* to do what it takes to get a great sound. When you put out an album it has to be 30 minutes of magic; it has to be worth $8.98."

The crusade by Bowen and his L.A. cohorts has born fruit. Budgets for top Nashville artists are way up, and album projects take weeks instead of days. On the liner notes for the *The Closer You Get* album, Alabama, for some obscure reason, listed their total studio time: 317 hours. (It helps, of course, when your co-producer, Harold Shedd, owns the studio, and you can be assured of multiplatinum sales!)

Digital Recording Technology

So what about digital? Is Nashville still hiding its collective head in the sand, or is it poised to jump on the bandwagon?

"On the *Billboard* country singles

chart of September 22 we had 42 singles," notes **Glenn Meadows** of Masterfonics disk mastering, and Master Technologies digital rentals. "Of those, 36 were [mastered] from JVC digital two-track. In cutting room #1, where I work, it's 80% digital now. Our overall digital business is up 30% from a year ago."

Over at Woodland, mastering engineer **Denny Purcell** reports that about 30% of his business is digital, either on his own Sony PCM-F1, or Woodland's Mitsubishi X-80 digital two-track system. All the other major mastering houses in town — Disc Mastering Incorporated, Custom Mastering, and Master Mix — either have in-house digital systems, or rent them regularly. Digital two-track mastering *is* running rampant across the country charts.

To demonstrate the growing impact that digital technology has had on country sessions, no less than six singles, mastered and/or recorded using Mitsubishi's X-80 digital two-track, were to be found on *Billboard's* Hot Country Singles chart during mid-September. The singles included Charley Pride's "The Power of Love"; Exile's "Give Me One More Chance"; "Your Heart's Not in It," by Janie Fricke; "Prisoner of the Highway" and "Still Losing You," by Ronnie Milsap; and "Where's the Dress" by Moe Bandy and Joe Stampley.


Photo by: Beth Gwinn

Producer Jimmy Bowen *(left)* with George Strait at Sound Stage Studio A, whose control-room equipment includes a Neve 32-in/16-out Model 8068 with NECAM automation and Studer A800 and A80 multitracks. The monitor system is a custom George Augspurger design powered by AB Systems, BGW and Crown amplifiers.

"All the early horror stories about digital have been put to rest by hands-on experience," claims Meadows. "I think it caught on suddenly here because Nashville, being a little bit slow to jump on the digital bandwagon, missed out on some of the fine tuning by the manufacturers. People in New York and Los Angeles burned out early with the problems they were having, and the way the early systems sounded. Also, we've been able to prove to people that it sounds better than analog, especially if you play it back four hours after the mix."

Master Technologies now has two JVC PCM processors out on rental constantly, with a third in-house for mastering. Mitsubishi has X-80 two-track rentals available through Digital Associates, and PCM-1610 systems are available through the regional Sony Professional Products office. And Sony F-1 systems are everywhere.

"There is a tremendous interest in two-track now, particularly in the F-1," says **Graeme Goodall**, southeast regional manager for Sony. "If nothing else, they are dabbling in digital, by using it for archival storage and for rough mixing. Some people are mixing to it and cutting from it, although we don't think that's a particularly good idea. Despite the fact that we don't think it has the error correction or the headroom, F-1s *are* being used."

One person who has been using the F-1 for more than a year is Rick McCollister, who has helped develop a method for cutting vinyl disks from the F-1 *without* using a digital delay unit to provide a preview signal for the cutting lathe. The technique involves the use of two F-1 units and two VCR transports: one for program audio. and one for preview. A blip is put on one tape three seconds before program, and a second blip 1.1 seconds (for a Neumann lathe) after that. The two VCRs are servo locked that distance apart, with one feeding the cutter head advance and the other the cutting amplifier. (This method has also been employed at both Woodland and Master Mix.)

Although both Purcell and McCollister are solid digital converts, they both consider that the technology suffers from imperfections, particularly regarding phase shift. McCollister has been using the Sony PCM-3324 digital multitrack extensively, and he's been doing some experimenting. "The 3324, with John Meyer phase correction, is indistinguishable from input except for a subtle difference in noise character. However, without phase correction, there is the characteristic change where the upper midrange and top is more pronounced and grainy."

McCollister has been working with the Sony 24-track in his basement workshop/studio, which was built in partnership with Bud Logan, producer for country artist John Conlee, and is considering a purchase. In the meantime he has been working on an interface between the 3324 and the F-1, to provide off-line storage. "I'm exploring the possibilities of off-line digital processing of signals. It's the same basic concept as the [NED] Synclavier and Fairlight in music generation, but in this case the music is already recorded. The idea is to take it off-line, put it into a computer, and manipulate the signal in a whole new way."

Glenn Meadows, president of Masterfonics disk mastering facility, and Master Technologies, a digital rental company that specializes in JVC digital products.

Photo by: Sam Borgerson

Although Sony and Digital Entertainment Corporation, the sales and marketing outlet for Mitsubishi hardware, report much more interest in digital multitrack this year — particularly in the rental field — system sales have not yet materialized. Nashville studios are willing to rent but not to buy for two related reasons: recording format standardization, and subsequent developments from Studer.

"There are too many incompatible formats," says **Glenn Snoddy** of Woodland. "Apparently they'll just fight it out in the marketplace." Although Woodland has purchased a Mitsubishi X-80 two-track and DDL-1 delay line, the studio is watching the multitrack fight from the sidelines.

"Response to the coming of Studer digital has been extremely positive," claims Joe Bean, southeastern region representative for Studer Revox America. "We have people who want to place orders; who are willing to put down money for product sight unseen. But it will still be only the high-end studios, the top 20%, who will invest in their own digital machines right off the bat."

Harold Shedd's Music Mill is among them. "We've been waiting for Studer to come out with their [DASH] digital machine," says Music Mill engineer Paul Goldberg. "That's been Harold's main concern. He's almost sure he would buy a Studer digital if it were available now. In the meantime we're renting the Mitsubishi and JVC systems."

Two studios in Nashville have decided not to wait for either Studer or

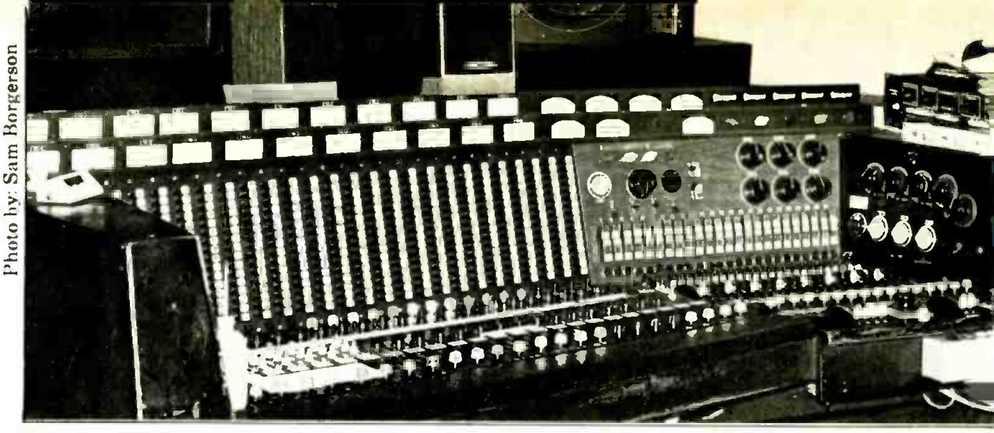
NASHVILLE IN THE EIGHTIES

a multitrack standard. Treasure Isle has owned a 3M M81 digital 32-track for nearly a year, and owner/engineer **Dave Shipley** reports that Emmylou Harris (another recent immigrant from L.A.) has been using the system, and gospel producer Greg Nelson is slated for three digital multitrack projects in the coming months.

"I mix differently when I use digital," says Shipley. "I notice I'm using a lot less EQ because I'm not wearing out the tape. It's quicker too, because you don't have to spend so much time trying to make it sound better; it sounds like it did when you cut it. Also we've been mixing to two tracks of the 32-track, punching in and out without having to do the whole mix over again." A nice added feature, since Treasure Isle's Trident console is not automated.

From this brief survey of the Nashville scene, we can assume at least some of the studios, producers, and engineers have pulled away from the backwaters of recording. And at least one studio, nestled in the rolling hills 15 miles from Music Row, has decided to catch the wave of the future.

Photo by: Sam Borgerson



Then and Now: the control-room interior of CBS Studio B — the "Quonset Hut" — dating from 1980, and a room that saw a lot of live country sessions.

Belgian Influence: The Castle

Throughout the Seventies, Belgian businessman **Jozef Nuyens** devoted considerable time to a second career as a musician, playing a blend of jazz and progressive bluegrass. His bluegrass connections led him to Nashville, where he found the musical atmosphere very congenial. Having moved his family across the Atlantic, he started looking for a good place to house a recording studio. A solid, live room seemed like a good idea, so he bought a castle. The imposing stone structure, despite its medieval facade, was actually built by a Chicago bookie/gambler (with supposed underworld connections) during the prohibition era.

In its first incarnation, the Castle housed a Harrison console and a single Studer A80 24-track. A few years later, Nuyens upgraded by adding a second A80 with a TLS2000 SMPTE Synchronization system for 48-track work. In 1983, digital arrived with the purchase of 3M M81 32-track and four-track units.

"We did several tests," says studio manager **Joseph Nuyens Jnr.**, "and we thought the 3M sounded the best when compared to the other two [available digital] multitracks. The others had some features we liked, but the best sound is what we go by here. Compared to analog, the difference in sound is remarkable; whether people want to face it or not, the digital just

Glenn Snoddy of Woodland Studios, a Neve/Studer-equipped facility that recently purchased a Mitsubishi X-80 digital two-track and companion DDL-1 delay line for use in its two disk-mastering rooms.

sounds *better.*"

After a few months of digital projects, Nuyens faced a problem anticipated — with dread — by other Nashville studio owners contemplating the switchover: previously unnoticed noise from an aging console and pre-digital studio wiring. So the old board left, the wiring was redone top to bottom, and a new 48-input Solid State Logic 4000E console brought in.

"When you have one [element of the] audio system that is pure," says Nuyens, "you have to change everything else to get it just as pure — everything has to change to get the best possible sound." Apparently the phrase "Good Enough for Country" is not comprehensible to natives of Belgium.

Several of Nashville's most progressive country/pop acts have visited the Castle, including Emmylou Harris for mixing, and Deborah Allen for Nashville's first 100% digital album. But the Nuyens are looking beyond country and Nashville, soliciting clients in New York and London by offering a setup for synthesized music production that is laid out much like some newer rooms in London. The Castle has six synthesizers, including a Yamaha DX-1 and DX-7, plus the new Fairlight CMI-2X.

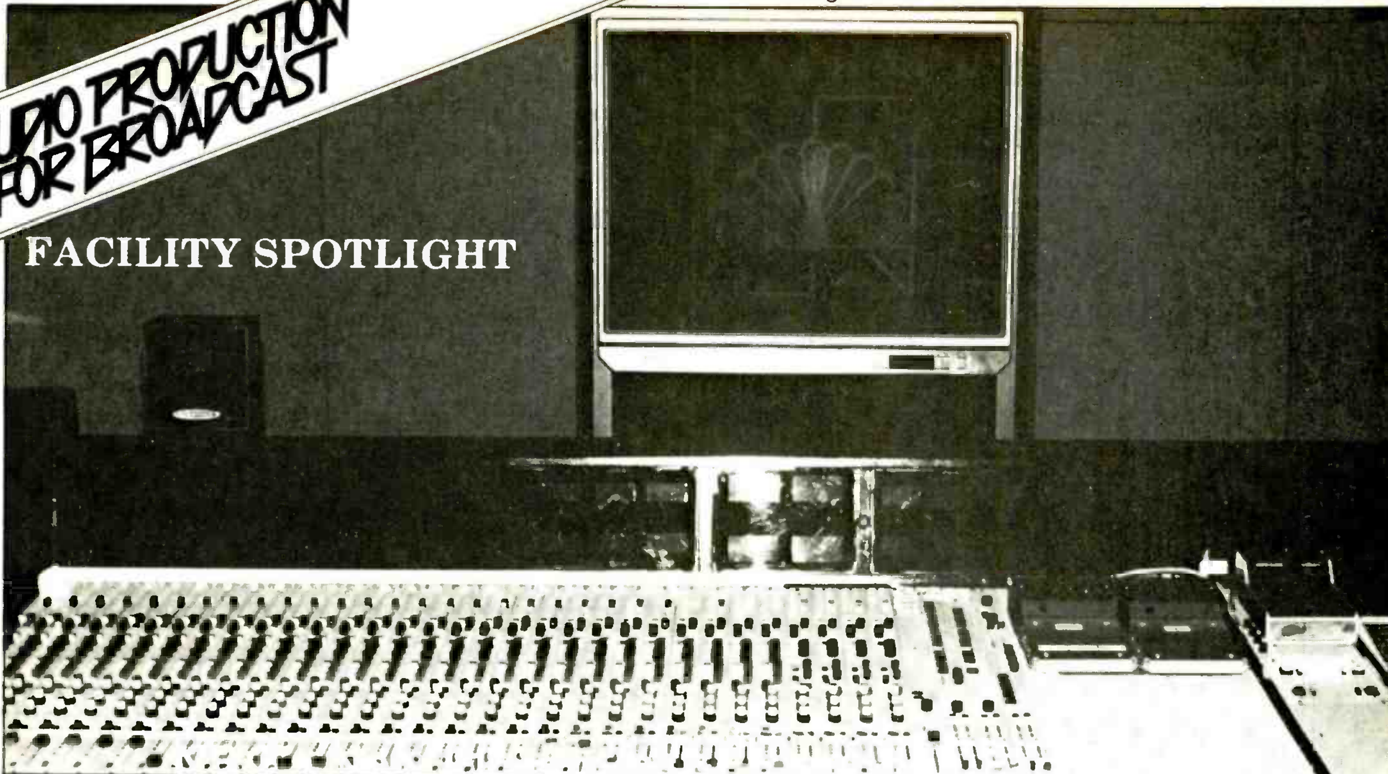
Even though it must be acknowledged that the Castle can hardly be thought of as a typical Nashville studio, it does mark the leading edge of an overall trend. If nothing else The Castle serves to remind Nashville conservatives — those who still insist "You don't need that for Country" — that the future is literally just down the road a piece. ■■■

The author would like to thank David Ross, of Music Row Directory, and Bob Millard for their help in preparing this article. All text was composed in Nashville on an eight-bit digital system.

AUDIO PRODUCTION FOR BROADCAST

FACILITY SPOTLIGHT

Quad-Eight/Westrex Coronado console in NBC PPS-1



Photography by Jeff DiPerna

Mix Magic, an Independent Video Post House

by Adrian Zarin

In today's expanding audio post-production industry, flexibility and diversity are cardinal virtues for any facility. The inevitable transition between the current requirements of television audio sweetening and the advent of Stereo TV will require that facilities be able to handle both types of work for some time to come. Post-production mixers, moreover, continue to shoulder responsibility for integrating program sources on film and videotape — a responsibility that is growing as feature-film production and Music Video continue to exert a growing and mutual influence on one another.

Two recently-constructed post-production facilities located on the West Coast illustrate the importance of flexibility and versatility. For PPS-1, NBC's new post-production studio located at its Burbank complex, a Live-End/DeadEnd™ control room and an adaptable CMX video editing system will afford the capability of meeting the changing demands of television post-production. For Mix Magic, an independent film/video tape post-production house, an acoustically "unbiased" monitoring environment and versatile equipment are the cornerstones of a growing reputation as specialists in the area of integrating film and video.

The NETWORK APPROACH: NBC's NEW FACILITY

Pulling together the diverse elements involved in the construction of a videotape post-production facility can be a monumental task, particularly when that facility is part of a vast television network complex like NBC's Burbank studios. The requirements of the many individuals and departments that will be using the studio need to be coordinated with non-technical, administrative considerations.

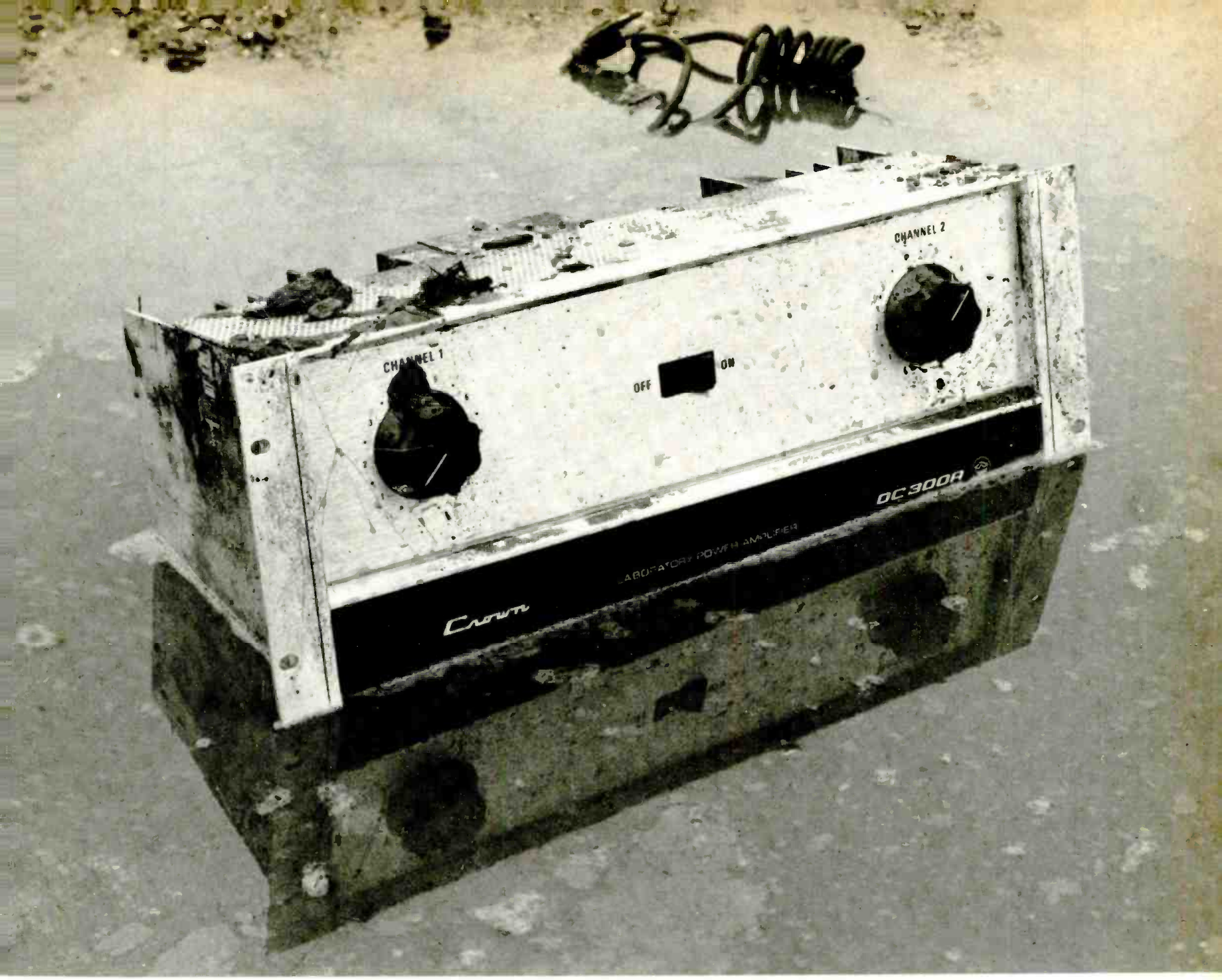
Design and construction of PPS-1 is no exception to the above generalizations. The task of assembling the facility fell largely to two men. NBC studio engineer, and regular *R-e/p* contributor, Peter Butt helped to integrate the equipment selected for PPS-1, while independent audio consultant Chips Davis was responsible for implementing an LEDE-style* control room design.

Construction of PPS-1 began in March 1983. By April 1984, the room was completed and began undertaking post-production projects. In June 1984, Roger Vonier signed on as PPS-1's senior mixer. He brings to his position some two decades of post-production audio experience, including tenures at Metrotape West, Pacific Video, and Greenlawn Video. Under Vonier, PPS-1 currently handles audio sweetening and mixing for a number of NBC shows, including *Punky Brewster*, *Days of Our Lives*, *One to Grow On*, plus assorted specials and sports programs.

According to Vonier, PPS-1 is not intended for only NBC in-house projects; he is, in fact, actively involved in making the industry at large aware of the facility. "My understanding is that PPS-1 was built primarily because NBC wanted to go after *outside* clients," he explains. "The facility is quite capable of doing many types of projects. Getting someone to use a new sweetening room can be a tough job, though. A producer or director gets very comfortable at the facility he's been using. It will take some time for word of mouth on PPS-1 to reach the industry, but out-of-house work is starting to come in already."

. . . continued overleaf —

*Live-End/Dead-End™ is a trademark of Synergetic Audio Concepts, the organization responsible for offering full certification to control rooms that have been designed according to strict LEDE principles. It is this magazine's editorial policy to refer to rooms that feature LEDE design principles as being "LEDE-type" environments, until they have been certified by Syn-Aud-Con —*Editor*.


CHANNEL 1
OFF
ON
CHANNEL 2
DC300A
LABORATORY POWER AMPLIFIER
Crown

FACILITY SPOTLIGHT: NBC PPS-1

Equipment Selection

Some of the equipment selections for PPS-1 were made on the basis of NBC's experiences at Vonier's former home base, Pacific Video. "NBC picked Pacific to do all of their network promos while they were building a sweetening room of their own," the engineer reports. "At Pacific, we had a CMX 340XP [video editing] system and Ampex MM-1200 24-track tape machines. When NBC came up with their room, they went for the CMX/Ampex combination as well —they were looking for a system that worked well, so NBC followed through on it."

PPS-1's complement of audio tape machines includes an Ampex ATR-124 24-track, an MM-1200 16-track (which is used as a backup multitrack) and ATR-Series two and four-track machines with changeable head stacks. A Technics Model RS-1620 transport handles any additional quarter-inch work that might be involved in a project. All of the Ampex ATRs are CMX-controlled.

"The Ampex machines are easy to work with from the standpoint of system design," indicates Peter Butt, "because they respond well to interface with the CMX editing system. The CMX has a 71-pin Winchester interface port for head control. It's generally all TTL control levels with a 9,600 Hz frequency-controlled capstan."

From an operational standpoint, Vonier also feels that the CMX was the best choice for PPS-1. "I've worked with the Adams-Smith and EECO synchronizers," he comments, "and I've found that the CMX is a lot more complicated; but along with that it's more versatile. It's more than just a synchronizer — it can do audio editing as well, and comes in handy for conforming tracks. We can feed in an edit decision list and do an auto-assemble to conform [the master audio tracks to follow video edits], which works out very nicely. I've had to hand-conform a few projects, and it's like a jigsaw puzzle; it takes *forever*."

The consoles installed in the new studio are, in fact, boards that the network already had in storage at the complex. The main console is a 40-input/24-buss Quad-Eight/Westrex Cornado, on which Butt has made some modifications. "I've added half a Farad of capacitance to the ±28V and ±15 rails," he explains. "There is an eighth of a Farad on each of those four rails, so it comes out to half a Farad of capacitance, which is enormously useful. I also removed a lot of the ceramic capacitors from the program chain. I modified the monitoring system with an external switcher, and added a lamp for indication of record function on the meter bridge of the console and at various locations in the room. There's an indicator above the sound effects console, and in [PPS-1's Foley] studio. This way, the sound-effects man knows when he's on, and the talent knows when he's on. If we're doing a tight insert, he knows exactly whether he's made it or not.

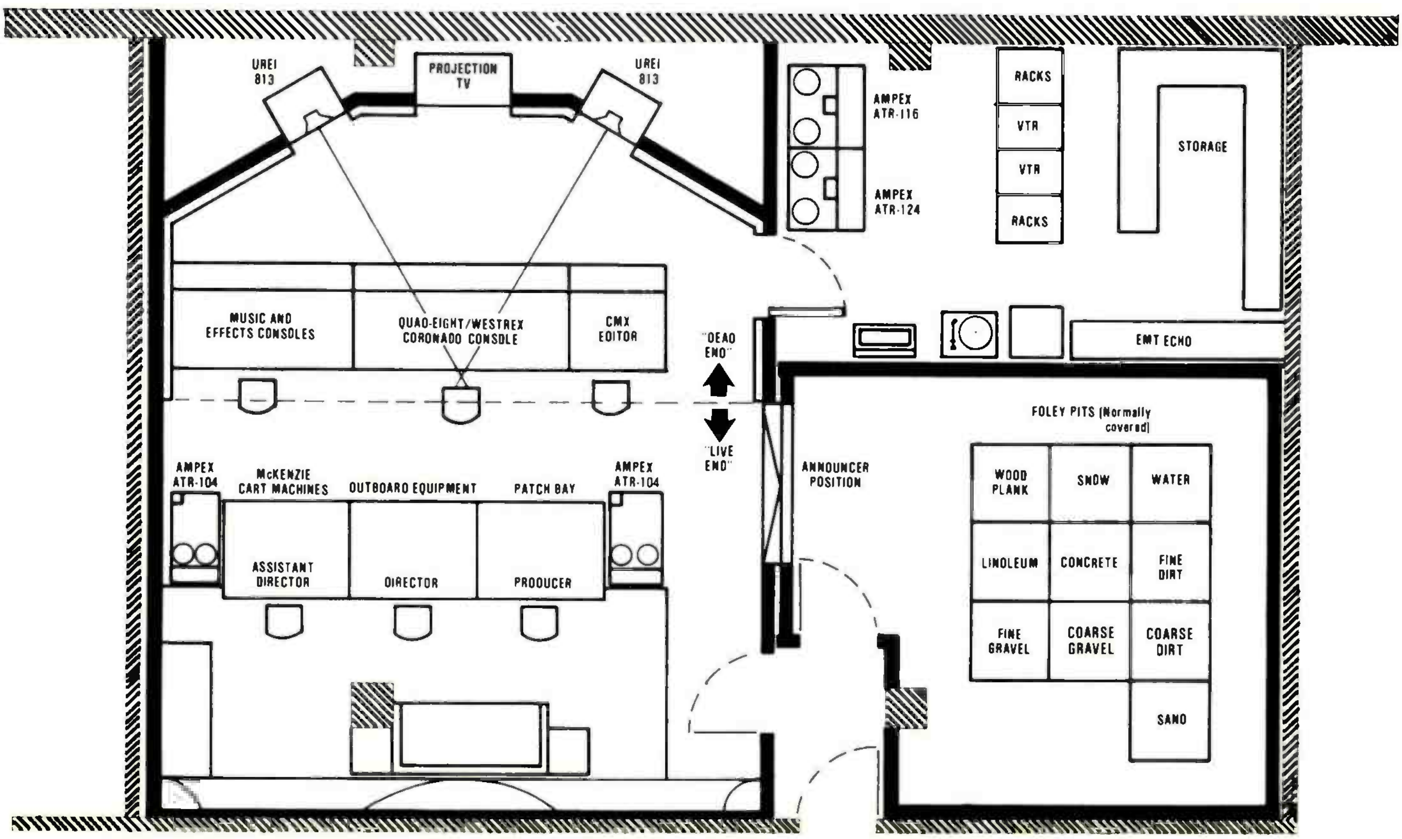
"Along with these modifications, I also added a relay switcher to the console for mono and stereo compatibility checks, and for comparisons between large and small speakers."

The Coronado is augmented by two, 16-channel Yamaha M-916 consoles that are used by PPS-1's chief effects mixer, John Burris, and the mixers who work under him.

"Using the Yamaha boards," Vonier explains, "John takes care of the sound-effects and music cues, and feeds them into my console. The Yamahas are, in effect, submixers."

Control Room Design

While Butt was working to integrate the equipment designated for the room, as described above, designer Chips Davis worked with the physical space and layout requirements that NBC had established for the new facility. Davis was engaged to implement a Live-End/Dead-End design, the basic premise of which, as many *R-e/p* readers will know, is to create an anechoic monitoring environment that does not color the sound. This is achieved by controlling early reflections from the front wall of the control room (which can cause broadband anomalies in frequency response due to phase cancellation/addition), and by using diffusion to eliminate the influence of the control room's rear wall. Achieving such an environment involves the creation of a



A CMX Editing System in PPS-1 *(left)* provides synchronization and machine control of various multitracks, stereo and NAB cartridge players used during post-production sweetening. It is seen here beside the full-function remote control unit for the facility's Ampex ATR-124 multitrack. The main outboard equipment rack houses dbx Model 165 compressor limiters and 900 Series modules, Eventide H949 Harmonizer and SP2016 digital processor, EXR Projector and two Exciters, four MICMIX Dynafex noise reduction units, and several cartridge machines.

FACILITY SPOTLIGHT: NBC PPS-1

symmetrical inner shell, with the length, width and height of the control room carefully chosen in a precise proportional relationship established by computer application of LEDE principles to the selected space.

Realizing an LEDE design in PPS-1 was particularly challenging, its designers concede. The space allocated by NBC for the new facility had more than its share of peculiarities, including a ceiling that lost roughly a foot in height toward the rear of the room, and a large ceiling support post, also located to the rear.

"The room was a bear," Davis confesses. "Apart form its physical dimensions, I was given a set of prerequisites regarding what equipment went in the room, and how NBC wanted it positioned. Ceiling heights and other dimensions were 'givens,' so it was a matter of varying the lengths and widths of the room to make things come out as optimally as possible within the space.

"Another thing that gave me some problems was that they wanted two rows of control points [a mixing area and a 'producer's area' behind it] instead of one. Fortunately, I had some latitude on where the front mixing desk could go. That's what I kept adjusting in order to get things to work out within the parameters I had. I determined by computer modeling what was happening with the modal fields in the room, and where the mixer should go."

But even with the basic proportions and positions established, Davis found that that equipment layout posed further problems. "I was worried about the rack of (outboard) equipment behind the mixer," he explains. "A direct reflection coming from there within less than 10 milliseconds would cause a big problem. It would end up in the total sound field instead of the *direct* sound field, which is where it should be. I managed to position things so that the main mixer doesn't hear the reflection, however.

"Unfortunately, the sound-effects mixers end up getting it, which was unavoidable. One effects mixer is right against the side wall and the other is about three feet from the wall. But as for the main mixer, who is going to have to make the most critical judgements anyway, I was able to make everything come out excellently."

With its emphasis on achieving optimum acoustic response through natural means, Davis' design philosophy is one that is strongly opposed to the use of room equalization. PPS-1 is no exception. Although its generous complement of effects hardware includes a UREI Model 537 graphic and Model 546 parametric equalizer, there are *no* room equalizers of any sort.

"You should only use equalization to paint your sonic picture," Davis comments. "You shouldn't depend on it to make the room respond correctly."

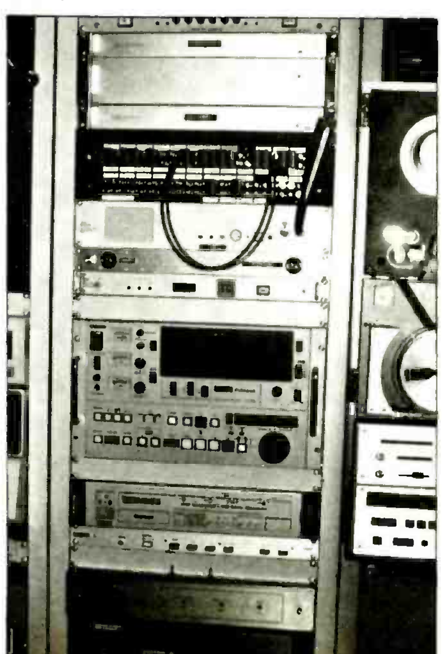
"I'm very much against room equalization as well," adds Butt. "People get into frames of mind where they're slavishly devoted to the flat curve. With a third-octave EQ, you can get a flat curve, but a lot of times it sounds *awful*. The reason is that you have these incredible deviations from the mean of your equalization curve. To kill a pitch, while bringing up a notch that's immediately adjacent to it, [means] you have to go to some extremes."

Monitor and power amp selection also fell to Davis, in his capacity as the room's acoustic designer. He chose UREI 813-B Time Aligns™, dampened and mounted according to LEDE practices, UREI and Crown Delta Omega™ amplification is utilized. The primary UREI/Crown monitoring system is backed up by Auratone Sound Cubes, driven by BGW Model 250 amps.

"When you get into a controlled environment," says Davis, "you *must* use 'phase-coherent' monitors. Otherwise, you're back to the same problems with phasing and group delay that you designed the room to avoid. Why should a speaker system [with its own components out of phase] add something that isn't really there?"

PPS-1's audio chain is complemented by an array of video equipment that includes a Sony BVH-1100A one-inch VTR and a ¾-inch Sony BVU-800 U-Matic, both of which are under CMX control along with the audio recorders.

The machine room houses several ¾-inch U-Matics, video switches, and and RCA one-inch C-Format VTR.



Audio Sweetening

"When we get a show in," Vonier

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FACILITY SPOTLIGHT: NBC PPS-1

explains, "we generally get a one-inch master. We put it up on our one-inch machine and do a layover, at which time we copy the audio with timecode onto our 24-track machine and make a ¾-inch video copy with timecode from the master burned into the picture. Then, during the sweetening and the prelay, we run the ¾-inch and the 24-track together, along with any four-track material we have, or audio sources on other tape machines. By handling things this way, we avoid shuttling the one-inch master back and forth and chewing it to bits."

PPS-1 has access to NBC's sound-effects library, which Vonier and his mixers will frequently draw upon in sweetening a show. "Before we do a project," he explains, "John Burris usually talks to the producer and director. He finds out what is needed and secures it from the sound-effects department." Many of the effects are stored on audio cartridge, and PPS-1 is equipped with an ample supply of conventional NAB and early-style playback McKenzie cart machines. The mono McKenzie machines, which incidentally, utilize quarter-inch tape that can be loaded by the operator, are fired by a Melkuist (now Audio Kinetics) events controller, while the NAB machines are under CMX control, along with the various reel-to-reel audio and videotape machines. With all of these disparate sound sources, even in-house projects at PPS-1 often involve the same sort of compatability problems faced by independent post-production houses.

"Although there shouldn't be a problem with standardization," Vonier comments, "there often is. Every area and department within NBC has its own techniques. We are accessing program material from ¾-inch cassette or one-inch iso reels that might have bad timecode on it. If the production sound man has been kind enough to think further down the road, he's given us some on-location sound we can use. Other than that, it's just a matter of searching and seizing all the elements we can from ¾-inch, audio cartridges, the McKenzies, or whatever. We put them together as best we can. This is where the CMX comes in very handy as well, since it enables us to fire up to eight different machines."

Sweetening a show at PPS-1 might even involve some on-the-spot Foley work. As a result, the facility has its own eight-surface Foley area adjoining the control room.

"If we're cruising along and come to a spot where there's too much noise," says Vonier, "we might find ourselves doing away with all sense of physical movement in the course of doing away with the noise — the scene becomes *dead*. So whoever is available — usually it's John Burris — will just go into the Foley room, pop a lid on one of the surfaces, turn on a [video] monitor, and we can put the sense of movement back into the scene. Although we don't have a wide-screen [video] monitor anywhere in PPS-1, we can handle ADR much in the same way by grouping people around the monitor in the Foley area."

Once sweetening of additional effects and Foley is completed on the multitrack, Vonier does a three-track relay (music, dialog and effects) to open tracks of the 24-track. The one-inch edited master is the locked to the 24-track machines for layback of the finished audio. Vonier prefers doing his prelays right to the 24-track, rather than using an intermediary four-track, because it enables him to make final adjustments to the mix during the layback.

"Mixing is a fluid situation," he explains, "A lot of times, you wind up with a fantastic mix — except for one or two little things. You can make a note and adjust those tracks during the layback rather than do a whole new mix. Doing everything on 24-track gives me a lot more flexibility. There's no sense in ruining a great-sounding mix just for that one little subtle thing."

Vonier reports that, although the facility has full stereo capabilities, most shows at this time are leaving PPS-1 in mono. Along with the finished one-inch master, however, Vonier generally finds himself making several other copies of the audio simultaneously during the layback.

"With the Quad-Eight," he says, "we can split off a lot of different mixes all at the same time. People usually want a two-track, quarter-inch copy, an audio casette, a VHS copy, etc. We can usually accommodate that quite easily."

Pictured left, from top to bottom: Senior engineer Roger Vonier at PPS-1's Quad-Eight/Westrex Coronado console; sound effects mixer John Burris in Foley re-recording area of the PPS-1 facility; acoustic consultant Chips Davis, who provided the LEDE-style design for the room, and selected the UREI 813-B Time Align monitor loudspeakers, plus UREI and Crown Delta Omega amplification; and Peter Butt, who is in charge of audio maintenance for all post-production rooms at NBC's Burbank complex.

40-input Trident TSM console at Mix Magic

FILM/VIDEO HYBRID: MIX MAGIC

Although it has only been in its present home since February 1984, Mix Magic is actually two years old. The audio for video/film service was founded in 1982 by Jim Corbett, a former film mixer for Todd-AO. Wanting to strike off on his own, but initially unable to purchase a state-of-the-art studio, Corbett came to an agreement with Motown-Hitsville in Los Angeles. With the record industry slump at its most critical stage, Motown had a studio that wasn't being used to its fullest capacity so this room became Mix Magic's first headquarters.

The setting proved ideal given Corbett's plan for Mix Magic, which was to apply the latest audio technology from the record industry to the special problems and requirements of audio post-production on film and videotape. "I wanted to experiment with a hybrid approach, integrating the music, video and film industries," he recalls. "Not many people were doing it at the time. So what happened was that we became problems solvers —audio consultants for a lot of the special problems that come up in mixed media projects."

The idea turned out to be a timely one, since the Music Video boom was then gathering momentum and generating more that its share of media integration problems. One of the clips that was brought to Mix Magic's Motown headquarters for a last-minute save was Michael Jackson's "Beat It."

"It had a major problem," Corbett recalls. "The two-track stereo audio on [mag] film did not match the mono tape copy of the song that the picture editor had cut to. We ended up varispeeding their original two-track master of the song to a half-inch, two-track running at 30 ips with timecode. It took about four trys, as I recall, to get a real close lip sync. The next morning, we had to bring in the CBS Records people to find out whether or not they would accept the VSO quality; the quality is slightly altered along with the pitch when you VSO a tape. Basically, they were in love with us for getting the picture and the audio in sync. So that afternoon, Red Car Editorial brought the soundtracks back. With Michael and Quincy Jones, we sifted through all the sound effects, and put together the complete audio track for the clip."

Music Video played a role in establishing Mix Magic's reputation and continues to account for a large share of the company's activities, and as everyone knows, videos were also a catalyst in revitalizing the record industry. As a result, the Motown facility became extremely busy once again, and the time was right for Mix Magic to set up its own permanent base of operations. Corbett was especially interested in securing a studio with a central Hollywood location, convenient to people from the record, video and film industries. A building in central Hollywood fit the bill perfecty, and became the present Mix Magic.

Meanwhile, Gary Fradkin had signed on as the facility's chief engineer; initially he became involved with the company in his capacity as a staff engineer at Motown, and eventually found himself a full-time part of the Mix Magic operation. Together with Corbett, Fradkin chose Brian Cornfield of Everything Audio to design the new Mix Magic mixing suite.

Facility Design Brief

The duo wanted a room based on the Motown control room that had served

DESIGNER'S NOTEBOOK:
Differences in Facility Designs for Television Audio Post Production and Record Production

by Brian Cornfield, president Everything Audio

The following article represents a brief overview of the main differences in the design of a TV post-production room, as opposed to a record production studio.

In our last two design projects for Mix Magic, Hollywood, and Master Control, Burbank, California, Everything Audio used #703 fiberglas within and on the outside of all acoustic walls to create a pleasant sounding, as well as acoustically accurate, audio room with a minumum of cost, time and space.

Both types of room have to meet certain audio performance standards, namely:

1. A stereo signal monitored in mono must not phase cancel, or otherwise distort the audio signal;
2. The left and right monitor loudspeakers must track each other, and not vary in overall frequency response by more than 1 dB;
3. The high-frequency response of the monitoring system must be within ±1.5 dB along the entire console listening area;
4. The frequency response of the loudspeakers must be linear over the entire listening spectrum;
5. The loudspeaker system will not require any additional equalization; and
6. That the rear listening area falls within ±1.5 dB frequency response tracking of the main monitoring environment.

Special wide-band traps, as well as custom low-band absorbers, were incorporated into the design of both rooms. Ceiling traps that contain lighting to improve the visual appeal of the rooms were also used.

Many acoustical systems are used in both types of rooms because of the need for high quality audio; these systems include:

A. Hanging ceiling traps to absorb high frequencies;
B. Loudspeaker wall traps to absorb mid and high frequencies;
C. Side and rear traps to absorb high and mid frequencies;
D. Low-frequency diaphragm systems built into front, side and rear walls themselves;
E. Noise barrier designs built into the front, side, and rear walls; and
F. Mechanically isolated subsystems used to decouple the floor system from the wall system, and the wall system from the ceiling system.

The most significant differences when dealing with TV post-production sound, as compared with record production, is the absence of a window in the control room, and the need for a highly accurate secondary listening area behind the mixing console for producers, editors, and other non-audio personnel. Other areas of difference involve the audio monitoring system. The size and quality of the speakers in both types of rooms have become more and more similar, due principally to the advent of Music Video promotion. Both of our latest rooms use high quality monitor systems: Mix Magic is equipped with UREI Time-Aligns, while Master Control uses JBL Model 4435 monitors.

Talent performance areas are also quite different. While a recording studio usually contains enough space for 20 to 30 musicians and their equipment, the usual requirement at post-production studios is for a voice-over booth in order to handle dialog recording.

Another area of difference is the location of the recording and playback machinery; construction and design allowances have to be made for these changes. In TV post rooms, the machinery usually moves from the floor of the control booth to a separate room. The type of machinery also changes, and the conventional audio tape machine is supported by magnetic film recorders and dubbers, plus videocassette as well as reel-to-reel videotape machines. All post rooms also include NAB cartridge machines and record players for sound-effects playback, while some facilities also are involved in the newest technology for storing sound effects in computer memory.

Design Considerations and Solutions

In both types of room, the main areas of concern are sound containment; sound conditioning; maximum space utilization; visual beauty; and cost. For these reasons we produced a layout for Mix Magic that has, as its first consideration, the ease with which an operator can use the room and all of its equipment. Secondly, we concerned ourselves with traffic patterns, in an effort to minimize time spent moving to and from areas within the room, as well as other areas in the facility. Side walls were designed

FACILITY SPOTLIGHT: MIX MAGIC

as Mix Magic's original headquarters. They wanted to preserve the octagonal shape of that control room, and continue to work with UREI Time Align monitors, along with the usual Auratones and three-inch television reference speaker. However, the new facility, which only allowed for a relatively small mixing room, imposed a few limitations of its own.

"We needed to get a lot of things into that small space," Corbett recalls. "We needed an area in the back for the producers, a mixing area in front of that and a near-field monitoring situation using the UREI Time Aligns. We only had about 19 feet in which to achieve that. We had a ceiling problem in that, given the way the rooms were built, the control room ceiling is only 14 inches below the roof of the building. We had a cement floor underneath, and we didn't want to change that. The control room was located near the bathroom, so we needed good isolation. And we also had to allow space for a voice-over booth."

"Acoustically, we needed a combination of a lot of different things," Fradkin adds. "We wanted to be able to listen in a record environment: flat out, with full frequency response. Also, for film dubbing work in a small room, we needed to be able to create an N-curve [the standard SMPTE room EQ curve for small rooms with volume below 5297 cubic feet, and similar to the well-known full-range X-Curve], and also an Academy [monitoring] curve.

"The basic room design is based on Live End/Dead End principles: the only solid wall is in the front, and everything else is soft surface. It enables us to get as close as possible to a flat response without initially using any EQ. We can then add [White Instruments room] EQ over it to get the N-curve and Academy curves we need. That was a major consideration, especially with the near-field monitoring in a small room like this. Now, this one room is basically set up to operate as three different rooms."

Equipment Complement

A Trident TSM console for the new facility was purchased from Tom Jung, who formerly had used it at Studio 80 in Minneapolis, before he relocated to A&M Recording in New York. The TSM has 40 inputs, 24 group outputs, and 32 monitor channels.

"It's an early Trident TSM, Fradkin says. "The circuit boards are all Issue #1, in fact. They needed a little work when I first got hold of them — I had to modify the group busses and bring those up to date a little bit. I also modified the way the console meters the outputs and changed a few switching functions, altering the way input signals play back through the board.

"The good thing about the Trident [TSM] is that it has off-line [or split] monitoring, as opposed to in-line monitoring, which made it a lot easier to adapt for post-production work."

"Basically we've been able to take what I think is a clean, state-of-the-art music console and adapt it for post-production work," Corbett adds. "The four-band quasiparametric EQ on the Trident is fabulous for this kind of work; it enables you to tune right in on frequencies you want to emphasize."

Corbett and Fradkin opted for a virtually all-Otari complement of tape machines for the new studio, including an MTR-90 24-track and MTR-10 with four-and two-track headblocks, and resolver unit. The only non-Otari tape machine is a Technics quarter-inch, two-track, which is used for playing in various music sources, including the audio tracks for music videos. Such tracks generally are transferred simultaneously to the Otari four-track (with timecode) and to mag film stock. The decision to use Otari machines was also based on Mix Magic's original Motown headquarters, which also was Otari-equipped.

"We liked the Otari machines at Motown and decided to stick with them,"

DESIGNER'S NOTEBOOK — continued . . .

using #703 fiberglas, sheetrock, resilient clips, and other materials, in order to contain and absorb unwanted and reflected audio energy.

Windows and doors in both types of rooms are areas of concern. In many cases, improperly designed and installed doors can be the point where most of the sound leakage occurs, which can be disturbing after going through such great lengths to contain sound energy. In these two installations at Mix Magic and Master Control we used three types of door: solid and hollow wooden types, and sliding glass. To work properly, wood doors must have a stop faced in rubber on all four sides. Glass sliding doors can be a very attractive addition, as well as providing an open feeling to a smaller room, but must be chosen very carefully to ensure that the manufacturer has provided good seals on all sliding surfaces; should this not be the case, extra seels must be installed.

Windows are not difficult to install properly, as long as three main points are remembered: 1) Provide a separate wall for each pane of glass to set in; 2) Use different thicknesses of glass; and 3) Use two different sizes to ensure that the panes vibrate as unsympathetically as possible.

Color and material selection of these two facilities was left to the owners; in many cases today's pastel colors and smooth fabrics are popular. Keeping in mind that colors can effect the mood and the feeling of a room, a pastel green and peach was used at Mix Magic, and a pleasant grey with burgandy accent selected at Master Control.

General Conclusions

The areas audio production for TV sound and records are moving closer and closer together, and the quality of both can only improve. Although there are some differences, as noted above, the same interest in high-quality audio production exists in both TV post production and the record-production industries. In working with both of these types of facility, it must be recognized that all the technical as well as creative information available has been used to create two excellent studios. We are glad to be involved in the technological and audio growth of TV post production and record recording in Los Angeles. □□□

FACILITY SPOTLIGHT: MIX MAGIC

Mix Magic chief engineer Gary Fradkin *(left)* and owner Jim Corbett at the facility's 40/24 Trident-TSM console, linked to Otari MTR-90 and MTR-10 tape transports via an Audio-Kinetics Q.Lock 3.10C synchronizer.

Fradkin comments. "We are now working with Otari on building a one-inch [video] layback machine for Mix Magic."

The Otari's were also chosen for their compatibility with Mix Magic's Audio Kinetics Q.Lock 310C synchronizer, another acquisition that results from the company's tenure at Motown, Fradkin relates. "The fact that Q.Lock came up with the first film interface with foot/frame conversion definitely influenced our decision to use it here. I also like the fact that we can download a CMX edit decision list to the Q.Lock. Audio Kinetics, moreover, worked with us very closely over at Motown to get the film interface working really well with Magna-Tech film dubbers, which seem to be pretty much the standard machine right now as far as the film industry goes."

Working on Mag Film

Mix Magic is equipped with four Magna-Tech 35mm mag recorders: two with four-track capability (expandable to six-track), and two mono. The facility also owns eight Magna-Tech dubbers: one with six-track playback capabilities, three with either three- or four-track playback capabilities, and the remainder single-track.

One of the things that Fradkin and Corbett are hoping to achieve through Mix Magic is to educate their clients as to the audio quality of mag stock. "We're planning to show the motion-picture industry that you can get very high fidelity out of mag," Fradkin comments. "The frequency response of mag recorders is about 30 Hz to 18,500 Hz using [3M Scotch] 350 stock. The additional headroom that mag has, and its response, are of great help. Hopefully, we can show people what mag can do."

"One of the things that this means is that we can do rock videos with the highest fidelity," adds Corbett. "We're talking about satisfying record producers who say 'I've got to hear that little bell,' which is out there at 12 or 14 kHz with overtones going well out into the 20 kHz range. We can roll their half-inch audio masters right onto sprockets, and not lose any generations. With high-fidelity mag, there are no longer *any* excuses for dull audio tracks."

Mix Magic's rack-mounted outboard equipment includes *(left)* a Valley People Dyna-Mite stereo compressor, a UREI Model 565 Filter Set, Dolby 360 unit with plug-in Cat. 43, Lexicon Model 200 digital reverb, Neve stereo compressor, and UREI 1176 limiter; and *(right)* Technics RS-1520 stereo reel-to-reel.

Mix Magic's film equipment is interfaced with three JVC 8250 and two 6625 ¾-inch videotape machines. In the interface system designed by Fradkin, the film chain is slaved to the video machines. "If you're locking your video to film, it's a problem," says Fradkin, "if you cannot reference your film machines to house sync or composite video sync. You can't get a good picture lock on video because you'll get roll. So it becomes necessary to lock film to video, and I see nothing wrong with that approach."

"Initially I wanted the sprocket machines to be the master and the video machines to be subservient," adds Corbett. "But because the video machines respond much faster than the film chain, it was impossible to do it that way, even though our film equipment runs at six times play speed, which is the maximum speed for film."

"In fact," Fradkin resumes, "one of the major factors for deciding on the JVC machines was that they run at a little slower speed than other videotape machines while they're shuttling. If you're shuttling back and forth on a 3½-minute spot, and you want to be able to catch small glimpses of certain things, other machines just run too fast to give you that ability. The JVC runs slow enough to let us do that, but fast enough so that it's not a problem. In fact, there's a little trimming you can do on it so that it doesn't run any faster than the film machines, where you're still limited to six-times splice speed."

Along with trailers for feature films like *Jaws III*, *The Amityville Horror* and *Psycho II*, plus post-production for television, Music Video still makes up a healthy percentage of Mix Magic's business. The facility has handled post-production for such prominent clips as ZZ Top's "Give Me All Your Lovin'," Lionel Ritchie's "Running With the

Night," and the Bette Midler/Mick Jagger rendition of "Beast of Burden."

Timecode and Frame Rate Anomolies

Fradkin and Corbett find that things haven't changed that much since the days of "Beat It," in terms of problems with Music Videos in post-production. Timecode, frame rate and resolve frequency anomalies still create problems on a regular basis. "There are a lot of people who are semi-knowledgable about timecode and transfers," says Corbett, "especially in the rock-video industry. I really wish that I could convey to some of these producers that they should touch base with a sound company *before* they shoot their video. What often happens is that they will take their two-track master, dump it to a little cassette machine, take the cassette out on location and lip sync to it. They then come in and expect their two-track to sync up with whatever they've got on film."

"There's also a problem of when to use [a resolve frequency of] 59.94 Hz and when to use 60 Hz," Fradkin adds. "For a lot of clips, people are shooting on film and then transferring to video, which can cause some confusion. Basically, if you're shooting on film, you should go with a 60 Hz resolve frequency. It's usually not much of a problem, but on a Jermaine Jackson video we did ["Dynamite"] they laid 60 Hz and then resolved to 59.94 Hz because they did the transfer at a video house. That made for some confusion later on when some of the mags were in sync and some weren't.

"Basically though, if you can find out what it is you have on tape [59.94 Hz or 60 Hz], you can always get around the problem. You can always resolve everything and then lay on your own timecode."

Apart from Music Video, as mentioned earlier, audio post-production of television shows and commercials makes up a very large share of Mix Magic's activities. Recent projects include a Christmas Special with Dolly Parton and Kenny Rogers for HBO Cable; *The Making of "Moscow on the Hudson"* documentary; network promos for ABC Television and KABC; plus numerous ad and jingle sessions for Honda (including the Devo spots), Coors, Dodge (one of which featured a stereo soundtrack), Miller and Budweiser.

In this area of their work, Corbett and Fradkin find themselves devoting a lot of energy to helping their clients understand stereo audio for television. "We're involved in an education process," Fradkin says, "where we are helping people prepare for the onslaught of Stereo Television, which is probably going to hit very fast and very hard next year; suddenly, these people are *very* interested in high fidelity audio. There are lots of questions and phone calls when a client is bringing in a stereo project — they are often very freaked out about what they need."

"There are a lot of problems with nomenclature at this point," Corbett points out. "People will say 'stereo' when they really mean multitrack separation of music dialog and effects. And then other people will say 'mono' and really mean 'stereo.' We often have to pull information out of people and find out what they really want. Once we've done that, though, it's generally easy to provide what they're looking for."

* * *

In designing post-production facilities during this period of transition in the audio production for broadcast, audio-for-video and film re-recording industries, the challenge lies in satisfying today's requirements, while still anticipating the industry's future needs.

For the designers of NBC's PPS-1 and Mix Magic, this has meant placing an emphasis on providing an acoustically neutral monitoring environment that can be adapted as needed. At the same time, the creators of both facilities have expended considerable efforts in combining audio for video/film equipment in a manner that allows the utmost flexibility in handling a diversity of projects. ■■■

AUDIO PRODUCTION FOR BROADCAST

THE GROWING IMPORTANCE OF HIGH-QUALITY STEREO AUDIO FOR VIDEO

Audio Production Techniques for Commercials, Jingles, and Video Sweetening

by Jim Dolan Jr., general manager, Streeterville Studios

Innovations in the communications industry are starting to bring many welcome changes in the quality of audio for video. Until recently, the treatment of television audio has been that of a second-class citizen. Unlike the radio, record and film industries, television has been content to broadcast in mono rather than stereo, and viewers have been forced to listen to their favorite programs through a miniature three-inch speaker that could just as easily been attached to their clock radio.

In the past, the justification for treating television audio like a secondclass citizen has stemmed from two key factors. First, since television was considered to be primarily a *visual* medium, it was only natural that the emphasis would be placed on images and pictures. Second, when the technology became available for stereo TV sound, broadcasters and manufacturers were unable to make use of it because the Federal Communications Commission had yet to decide on a standardized broadcast format. Time was needed by the FCC to test all the formats submitted by various manufacturers, a process similar to the one performed when the Commission had to decide on a broadcast format for color television. After a great deal of testing — and a great deal of behind-the-scenes politics — the FCC chose the Zenith-dbx process for stereo television audio.

Before long, many if not all television commercials and programs will be recorded and broadcast in stereo, receiver manufacturers will be putting the new technology into their sets, and television viewers will discover a whole new dimension to the medium.

The most obvious benefactor of enhanced television sound is Music Videos. Viewers soon will be able to watch their favorite musical group or performer on the TV screen, and have the sound quality they have become accustomed to when listening to their FM radio or stereo hifi.

But the benefits of Stereo Television go a good deal further than Music Video. Programs such as PBS' *Great Performances* series of classical concerts, which currently must be simulcast over FM radio, will no longer need to be broadcast that way. Programs in which music does not play a dominant role also will be affected; viewers will notice a higher anticipation when *Magnum PI* or *Remington-Steele* arrive to save the day due to the added dimension that stereo sound will be able to create. Better sound effects will enhance the excitement when *Knight Rider* jumps over a cliff or the *Fall Guy* is thrown out a window. Advertisers will be pleased to find their commercials having a stronger impact due to the enhancement that Stereo Television will bring to their pitches.

Even programs that use little or no music will benefit. With stereo sound, viewers will be able to hear Johnny Carson coming out of one speaker, and Ed McMahon from the other. And imagine a televised baseball game that will become even more exciting as the roar of the crowd and rally of the organ bring the television viewer even closer to the stands.

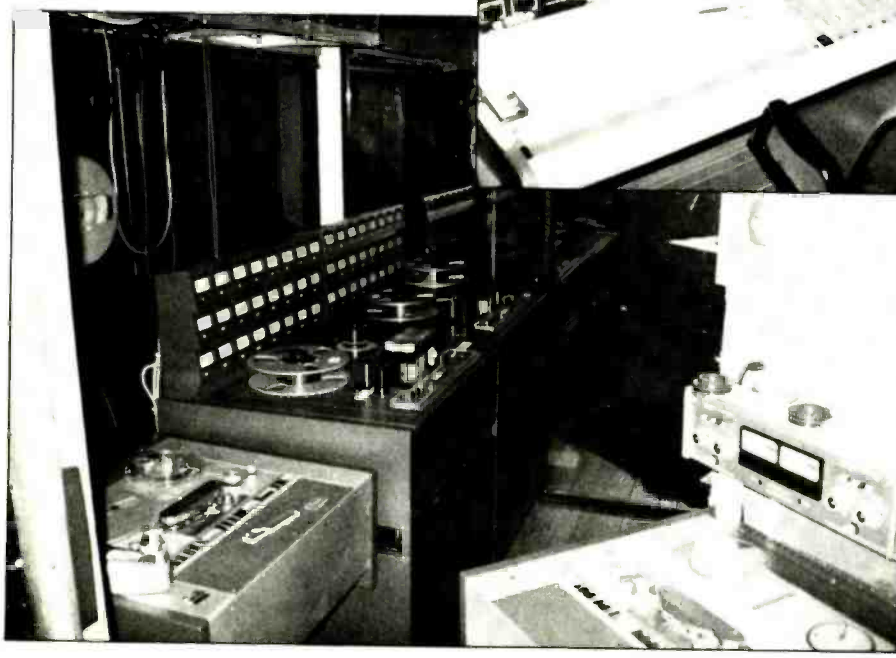
To the recording engineer, the long overdue advent of Stereo Television should be welcomed with open arms. No longer will we have to toil for hours trying to capture details that never made it to the final process. Now when we give our work that extra push it will be noticed on television; indeed, it will be expected. Stereo Television brings new excitement, new challenges, and new responsibilities to audio engineers. It is up to us to maintain the high sound quality that we put into records, films and radio when handling audio-for-video. It is also up to us to devise improved techniques for recording, mixing and layback of audio-for-video.

Television programs and, especially, commercials are generally made on tight schedules. While Stereo Television will enhance the demand for high-quality sound, it won't increase the time in which these programs and commercials are made. Therefore, it is up to us to develop fast and efficient production techniques. It is also imperative that we strengthen our relationships with our esteemed colleagues in video production. Although sound people have always had close ties with visual people, the new demand for high quality audio-for-video will greaten the interdependence we now share, and demand that we work even more closely to develop a premium product.

Despite the fact that Stereo Televi-

HIGH-QUALITY AUDIO FOR VIDEO

The control room of Streeterville's Music I studio *(right)* is centered around a 48-input Neve Model 8128 console with NECAM II, while the machine room *(below)* houses a pair of MCI JH-24 multitracks, two Studer B67 stereo machines, and an Otari MTR-12 four-track/half-inch deck.



sion is less than a year away from becoming a reality, most recording studios are still not prepared for the demands of audio-for-video. At times they find themselves flying by the seat of their pants, and end up costing the producer time, quality and money. At Streeterville, we have been handling audio for visuals since we opened for business in 1967, and have developed rooms, people and equipment especially for the needs of this specialist operation. Within a year, all our rooms will have audio-for-video capabilities, rather than just video playback.

Building on Multitrack

An example of our current operation is reflected in Streeterville's audio finishing process, which starts with a commercial producer bringing in a 24-track music master for mixdown. However, instead of optimizing the mix and transferring it to half-inch/four-track, or quarter-inch mono, with accompanying SMPTE timecode, the music mixed to another 24-track that maintains the mix's dynamics yet separates it across several discrete channels for later rebalancing, if necessary. This tape becomes the audio-for-video master and, once striped with SMPTE code, can be synched with a Sony ¾-inch BVU-800 videocassette, both controlled by an Audio Kinetics Q.Lock Synchronizer with Option 64 software. Now, any audio elements, including sound effects, announcer, etc., can be added to the master while viewing the picture in sync with the music mix.

Elements can be laid in from 35mm mag or quarter-inch reel-to-reel by utilizing the Q.Lock's auto start and record-enable functions. Announcers' voices, as well as musicians' performances, or synthesized effects, can be recorded to picture in sync, and laid straight to the audio-for-video 24-track. During the assembly process it is possible to replay the audio through a TV set, and more accurately assess the sound on a realistic replay system.

Having a television monitor is also useful during a mix for audio monitoring; our all-time favorite is a Sony Trinitron located in front of the mixing console, to enable the final adjustments of EQ, level, and ambience.

Once an effective blend of music, voice, SFX, etc., has been achieved for the producer, the client, and, hopefully, the engineer, the resultant stereo mix can be transferred to any, or all of the following formats: one-inch, Type-C videotape, ¾-inch U-Matic, half-inch/four-track, quarter-inch/ two-track or mono, 35mm and 16mm mag or even cassette.

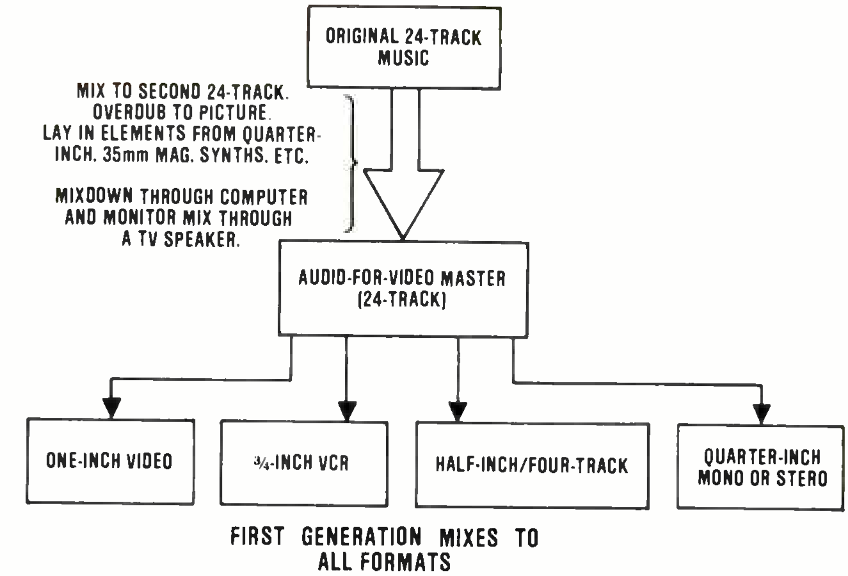
This 24-track format and sweetening process is vastly superior to the typical finishing process for a television commercial, which usually involves three-stripe 35mm mag containing three different sound elements. With access to 24 track you can spread out the recording over many more channels, the goal being, of course, to eliminate compromise and second guessing. For example, by utilizing this process I find that if a high horn section is getting in the way of a spoken line, I can deal with just that horn section.

A practical example of how effective the process can be is demonstrated by a recent project Streeterville recorded for a major advertising agency. All the sound had been layed out, and the commercial was in its final mixing stage. All of a sudden, the agency producer discovered that there was a hole between where the announcer's voice ended and the music picked up. Because of the availability of open tracks on the multitrack, we were able to add a synthesizer element and thus fill the hole. As a result, the client was able to save time and money in additional recording and mixing sessions.

As noted earlier, the A-K Q.Lock synchronizer with Option 64 software facilitates the laying in of elements via its Auto-Start and Auto-Record functions. These functions allow a quick and precise build up of the track, and can take advantage of laying in original audio from location recording, or by taping a pre-edited or pre-mixed effect

or voice track that the editor has created on 35mm mag. What matters here is not only which is best for the project from a quality standpoint, but also its cost-effectiveness in terms of time and money. It is very worthwhile to be able to shape the sonic qualities of your effects or your announcer to the rest of the music program while building a track. You end up with elements that work more effectively with the music, or vice-versa, which results in a more effective program.

During the final mixing of an audio-for-video project, a console automation system can be extremely useful, especially if you have just completed a 12-hour session only to find that a word from the soundtrack has to be removed (possibly for legal purposes) and the whole mix redone. A recent case in point: During production of a project for a major beer company, the agency needed approvals from several people, and it seemed that each of them had one thing or another they wanted changed. Normally, this can take a toll on an engineer but, because of our automation process, we were able to preserve the continuity of the mixes and complete the project in half the time. Also, first-generation mixes from the audio-for-video multitrack master can be passed across to any format needed for the job, the result being a higher quality product since fewer audio generations are required.



FLOW CHART FOR MAINTAINING HIGH-QUALITY STEREO AUDIO FOR VIDEO

Audio Layback to One-Inch

The final part of Streeterville's audio-for-video process is mixing down the completed tracks. There are two ways of handling this, either of which can optimize the process depending on the project's status, and the time available. One way is to mix down direct a one-inch videotape with color burst recorded already, which puts second- or third-generation audio back onto the one-inch mastertape via a second transfer stage. (In essence, at a video facility the master video is inserted right on the piece of pre-audio striped one-inch videotape.)

The second technique of recording remixed audio with video would be to lay back the audio right onto the final edited and color-corrected one-inch video master. Either one of these processes allow for top-quality audio to be put on videotape.

At Streeterville, we strongly feel that it is better to let us handle the audio layback, rather than take our finished soundtrack to a video facility. Video facilities, of course, are capable of doing the final audio transfer, but there are several reasons why it is advisable to let an audio studio do it, since most of the equipment available at video-houses is not designed for high-quality audio.

As may be appreciated, throughout the entire audio sweetening process, only three generations are used. Also, in cases where only one 24-track tape is needed (maybe the music is recorded over only a dozen tracks, for example,) we can go from start to finish using only two generations, which is a considerable improvement over the tradi-

FACILITY SPOTLIGHT: STREETERVILLE STUDIOS

Streeterville Studios, located in the heart of downtown Chicago, is comprised of 22 staff personnel and five studios. Services range from traditional multitrack music recording and narration/SFX production, to newer technologies such as satellite recording, computerized audio-for-video, as well as digital recording.

Streeterville general manager, Jim Dolan, Jr.

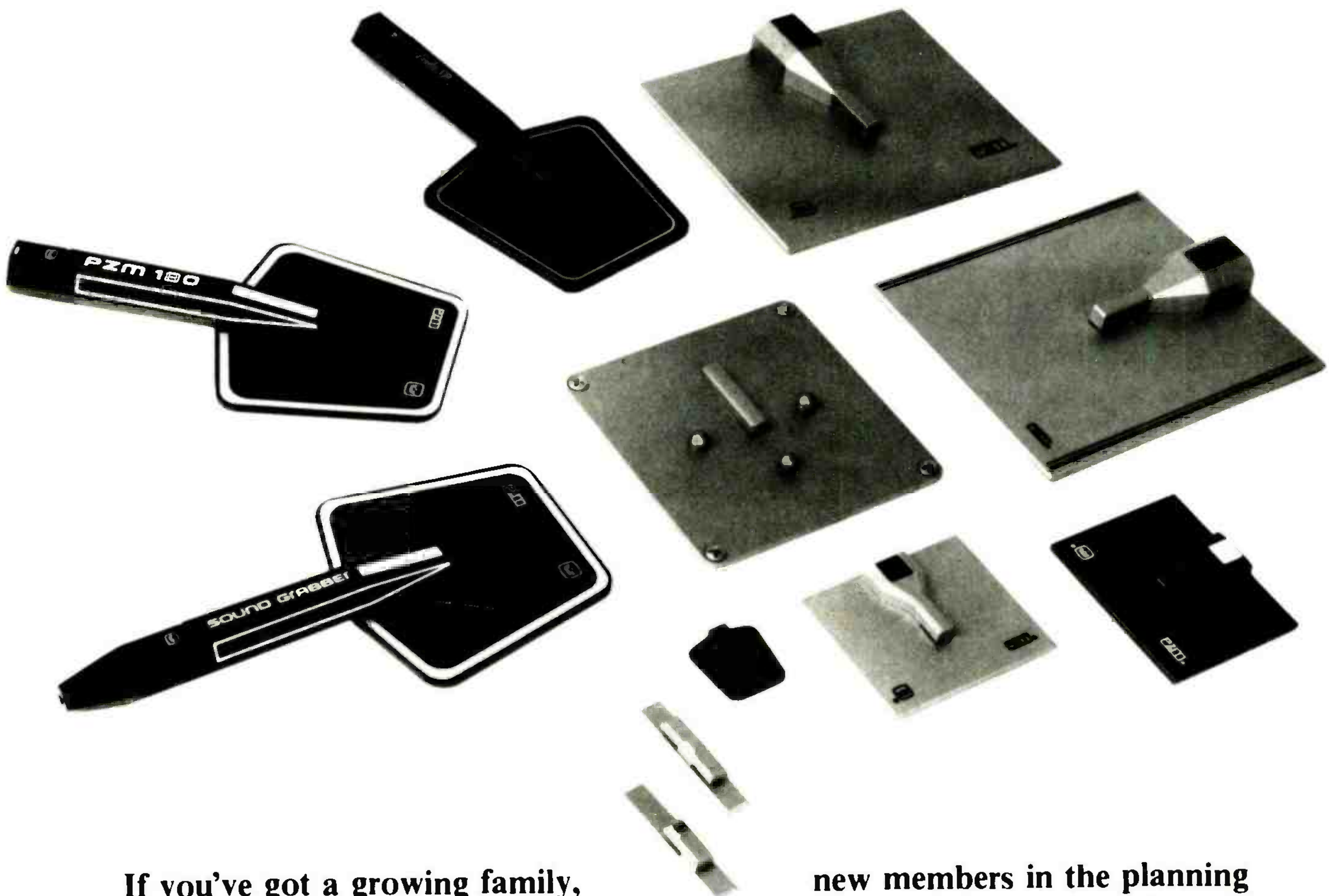
The company set up shop at its present location in 1969, in association with a television and radio production company, Shield Productions. The original facility consisted of Music I, the first studio designed for 16 or more tracks in Chicago, and Studios B and C, eight- and four-track production studios, respectively. Streeterville spent the next 10 years not only servicing its parent company, but also establishing itself as an independent studio in the growing record, advertising, and industrial markets of the midwest.

In 1979, based on that growth, Streeterville needed to expand its music recording capabilities, and chose George Augspurger and Jake Edwards of Perception, Inc. to design a pair of 24/48-track studios to be called Music II and The Suite. Along with the rebuilding of Music I in 1980, Streeterville was readied for the acoustical, technical, and aesthetic challenges of recording in the Eighties.

The complex also boasts a variety of support rooms, including a pre-production music and SFX room that houses various music libraries as well as up to date talent demo tapes; an extensive post-production room for 16mm or 35mm mag transfer, audio cassette as well as ¾-inch videocassettes dubbing; and a comprehensive quarter-inch mass tape duplication service that includes packing and shipping to radio stations.

Music I is described as a state-of-the-art, full-service 24/48-track music studio that

Family Portrait



If you've got a growing family, sooner or later you need a picture with everybody in it. It's a statement of family pride, and we humbly admit that we are pretty proud of this group.

There was a time when most people didn't recognize a Crown PZM® as a microphone - even when they looked at one. Times have changed. Billboard Magazine reports in their most recent brand usage survey that 37.5% of U.S. recording studios use Crown PZMs.

This sort of demand, multiplied by many other applications, has made the family grow, with new microphones tailored for new users. In fact, the number of new members in the planning process is larger than the number in the picture. Since a lot of our friends have only used one or two models so far, we thought we'd better introduce the family. The next time we may not be able to get them all in one picture.

Keep an eye on this family. Right now it's one of the newest and best. It just might get to be the biggest.

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tional 35mm mag format, in which nine or 10 generations of tape might be used.

From an engineer's point of view, the exciting aspect of this audio-for-video process is the fact that it allows the same mixer to follow a project form the downbeat of the music, straight through to the final audio layback, which ensures compatibility of the technical process from stage to stage. This way we have the opportunity to see our projects through, and thus develop a greater sense of pride and responsibility in our work. The process also streamlines the producer's job, because he relates his concerns to only one person as opposed to the standard four or five.

Music II's recording equipment *(above)* includes a Harrison 4032C console, MCI JH-24 multitrack, Studer B67 stereo decks, and a full collection of outboard signal processors. The Harrison board *(right)* features 40 input channels and 32 routing busses.

Towards the Future

With all the changes coming into the recording industry — Audio-for-Video, Stereo Television, digital technology — our responsibilities as studio people will be keener and more demanding in reference to decisions on equipment and processes. Our colleagues working in video and television facilities, who will be expected to facilitate stereo monitoring and precise audio playback performance, should be able to depend on our experience in those areas, just as we will have to look to them for video guidance.

Neither audio nor video studios are equipped to handle the new technologies alone. Of course, both audio and video studios should relish the new challenges that technologies such as Stereo Television bring, but we should always remember that neither of us have the capabilities to go it alone. We could try, but the resulting drain on finances and manpower would make it unwise. It takes good audio *and* good video to make a television production effective, and audio studios should be able to work in harmony with video facilities in the same way that sound and sight work together to bring the viewer his or her favorite TV program.

As for myself and fellow engineers, it is now up to us to rise to the challenge that Stereo Television will bring. Of course, recording in stereo is nothing new, but we will now have to produce high quality sound in, at best, the same time span. I refer, of course, to that old adage, "It's only a commercial. Nobody will hear it anyway." Those times are rapidly disappearing, if they haven't gone already.

As mixers we are all sensitive to the striving that must be done and the constraints that must be overcome for state-of-the-art in jingle sessions. Stereo Television will increase the need of commercials to be par soundwise with records and movies, and thus intensify the situation. In other words, while studios are turning out better commercials today than they were 10 years ago, they are still given a limited

FACILITY SPOTLIGHT — continued . . .

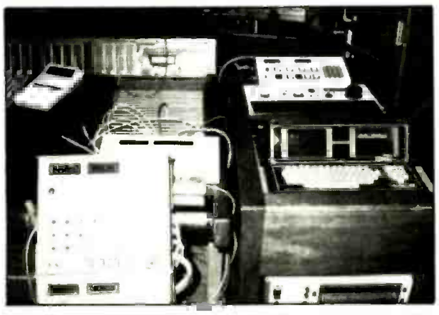
holds up to 40 musicians, and offers a variety of areas and surfaces from live to softer ambiences, plus an isolation booth. The studio package includes a Yamaha grand piano and Sonar drum set. Control room hardware includes a 48-input Neve 8128 console equipped with NECAM II; twin MCI JH-24 multitracks with Autolocate III; two Studer B-67 tape machines; Otari MTR-12 four-track; Studer B-710 cassette; Lexicon 224X digital reverberation with the LARC; UREI Time-Align 813B speakers powered by UREI 6500, Hafler 500, BGW 100B and Crown D150 amplifiers; Sony VO-5800 video deck; BTX 4500 synchronizer and 4600 controller; EMT 140 and 240 Gold Foil reverbs; MXR digital delays; Eventide Harmonizer; various Valley People Allison Gain Brains and Kepex gates; UREI 1176-LN limiters; Orban 516C dynamic siblance controller and 622B parametric equalizer; and Pultec EQP-IS equalizer.

Music II is a 24-track music studio that facilitates 25 musicians. The recording area is treated with hard and soft surfaces for desired ambience, plus a six- to eight-player isolation booth. Harrison 4032C transformerless console is linked to an MCI JH-24 multitrack with Autolocate III, and Studer B-67 stereo/mono tape machines. Effects include a Lexicon 224X digital reverberation with LARC, EMT 140 and EMT 240 Gold Foil reverbs, Eventide 1745 digital delay, Lexicon Model 92 digital delay, four Valley People Gain Brains and four Kepex gates, a dbx Over-Easy compressor/limiter, UREI 1176 LN and LA-4 limiter.

The Suite is a state-of-the-art finishing studio for record remix as well as computerized audio-for-video sweetening. The studio features an isolation booth that can accommodate up to 10 musicians. A Harrison 4032B mixing with Auto Set I computer automation links to a pair of MCI JH-24 multitracks, and Studer B-67 mono/stereo decks. Outboards and monitoring are similar to Music I and II.

Recent Projects at Streeterville include record sessions with Johnny Winter, James Cotton, Big Twist and the Mellow Fellows, K.K. Taylor, Crystal Wings, and Buckinghams; commercials for United Airlines, Bud Light, McDonalds, Tasters Choice, Sears, Coors, Taco Bell, True Value, and RCA; plus *My Roommate*, PBS' *American Playhouse*, and Johnny Winter's first music-video. □□□

The Suite *(shown top left)* **features a Harrison 4032B console with AutoSet I VCA-based automation, a pair of MCI JH-24 multitracks, and Studer B67 mastering machines. In the rear of the room** *(top right)* **are located an MCI JH-110 one-inch C-Format layback machine, and an Otari MTR-10 four-track/half-**



inch, and various outboard signal processors. The area to the right of the console *(shown left)* **houses an Audio Kinetics Q.Lock 3.10 synchronizer with Option 64 software, the AutoSet controller, MTR-10 remote, stacked JH-24 remotes, and remote for the lower, rack-mounted Sony BVU-800 ¾-inch U-Matic VCR.**

amount of time in which to do so. So all the more reason for studios to use recording methods that cut down on technical problems, allow the engineers to be more creative, and still finish the projects on schedule.

The fact that we can offer total audio service control and continuity from conception through the mix of a project, no matter what format or medium you are working in, makes the Eighties a very exciting time and place for a recording studio. It should be remembered that audio is just one piece of the television-production pie; as the demand for better sound grows, however, it is rapidly becoming a very important piece. ■■■

QUANTEC ROOM SIMULATOR

Reviewed by Bob Hodas

The Quantec Room Simulator is a sophisticated computer-based reverberation system that generates its effects by simulating rooms of variable size and acoustic parameters. The heart of the QRS system — and one that potentially separates it from the competition — is a 26-bit digital processor operating at a clock speed in excess of 20 MHz, coupled to two megabits of RAM. In other words, the device is extremely fast with massive memory! These factors, along with the unit's unique filter arrangements, eliminate many of the problems of transient glitches, "stutters," distortion and "roaring" that plagues other digital reverbs in this price class.

The QRS is a stereo-in/four-out unit, and produces over 10,000 reflections per second, giving the device a distinctly linear and realistic response. The four outputs represent four simulated "microphone" positions in a room, and may be used to create mono, stereo or quadraphonic effects. Since the first two outputs contain the first reflection parameter, three-dimensional effects are possible with just output #1 and #2.

Marshall Electronic, exclusive U.S. distributor of Quantec products, has further enhanced the unit for its domestic customers, by hand selecting several analog and digital components for the highest quality specs, matching, and sonic purity; units assembled in the U.S. and shipped by Marshall during the past several months now contain these and other upgrades.

As might already have been guessed, this reviewer found the QRS to be quite impressive, so let's take a look at the control functions and variable parameters to provide a solid picture of what the device is capable of achieving.

Since the QRS unit is labelled a Room Simulator, room size comprises the basic parameter: There are seven "rooms" available to the user, ranging in size from one cubic meter (with each successive room being one exponent larger) to 10^6 cubic meters — which relates in more familiar terms to spaces about the size of a garbage can, to one with a volume of 35 million cubic feet!

Changing the room size affects four simulation factors:

1. Delay time until start of reverberation;
2. Reverberation rise time of the room;
3. Reflection density of the echo;
4. Density and distribution of the natural resonances of the room.

While each of these factors is changed automatically when selecting different room sizes, there are seven additional, user adjustable parameters:

Reverb Time, whose limits are dependent on room size, but with plenty of discrete settings whose upper and lower RT_{60} limits are as follows:

Room Size	*Minimum RT_{60} at 1kHz*	*Maximum RT_{60} at 1kHz*	*Number of Settings*
1 cubic meter	0.1 second	1 second	21
10	0.1	2	27
10^2	0.1	5	35
10^3	0.1	10	41
10^4	0.2	20	41
10^5	0.5	50	41
10^6	1.0	100	41

Low-End Reverb Time, for which there are 11 settings between 0.1 and 10, the fundamental reverb time being reached at around 40 Hz. This setting is limited to a maximum scaling factor of 10 and a minimum of 0.1 times the reverb time selected for any given room, and is based on known properties of reverb to create a very realistic sound. Included in this parameter is a linear setting (specified "LIN" on the LED readout) for the most natural, even reverb, the other settings acting to prolong or curtail low-end reverb around the central setting.

High-End Reverb Time contains eight multiplier coefficients ranging from 0.1 to 2.5 times the RT_{60} selected for the room, with a linear setting also included as its central setting. High-end adjustments are centered around 8 kHz.

First Reflection Delay appears on the opposite side of the stereo image input, and can be used within the reverb program or independently as a discrete stereo echo. The parameter can be varied in 1 millisecond steps between 1 and 200 milliseconds.

First Reflection Level mixes the desired reverb level with the original (delayed) signal, and is variable in 1 dB steps from 30 dB to 0 dB of attenuation. (There is also a mute or "off" setting.)

Reverb Delay — while the room-size program already takes this factor

THAT BRITISH SOUND

BOTH TAPE MACHINES FEATURE: +4dbm IN/OUT • 15/30ips • Full-function 9 cue position remote-autolocator • Stand • 50% range vari-speed •

ACES TR-24: 2" 24 track Recorder/Reproducer **$19,950.**
(2" 16trk., pre-wired 24trk. available)

ITAM #1610: 1" 16 track Recorder/Reproducer **$11,950.**
(1" 8trk., pre-wired available)

ACES ML24: I/O console, 32 in x 24 buss, Integrated part-wired patch bay. **$17,025.**

ACES SM16: Split console, 32 in x 16 buss **$15,665.**

BOTH CONSOLES FEATURE: Fully modular • +4dbm IN/OUT • AudioFad long-throw conductive plastic faders • 5 Aux sends • 5 band switchable EQ • Input LED PPM'S • Stand • +48v phantom power • LED display (optional) • Two year parts warranty • Other frame sizes available • Many options available

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For additional information circle #97

QUANTEC ROOM SIMULATOR REVIEW

into account, an additional 200 milliseconds of delay may be added before the onset of reverb; the parameter is adjustable in millisecond steps.

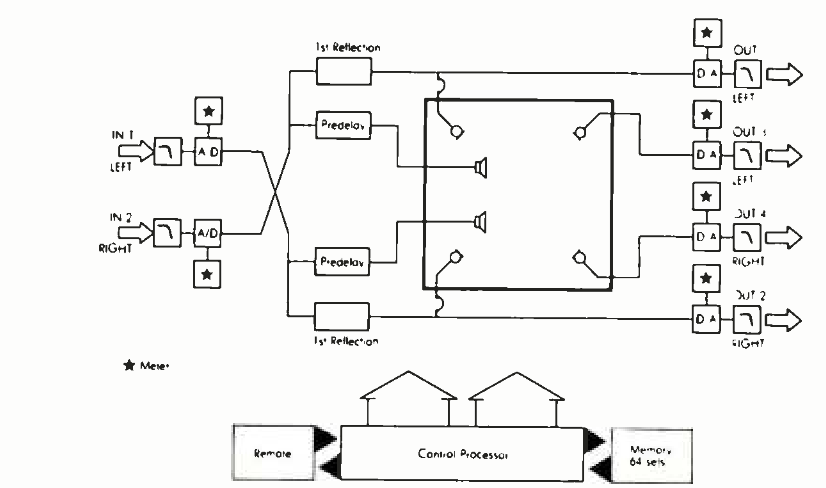
Reverb Level is also variable in 1 dB steps from -30 dB to 0 dB. The control setting can be useful in defining with great accuracy the amount of reverb mixed in to the original signal, and is much more controllable than channel faders when switching quickly between programs.

In addition to the control parameters are two nice special effects, "Freeze" and "Enhance." Set to Freeze mode, we have a room with a reverb pattern of infinite decay time; the unit selects an imaginary room in which all frequencies are 100% reflective, and there is never any absorption — the room loads up and becomes pure reverberation. One can continue to add new sounds into the Freeze mode, building up chorus-type effects or creating whole new sounds with various noises or instruments. (If you saw *Buckaroo Bonzai*, the rocket engines were created with this mode.)

[According to Steven St. Croix of Marshall, the Quantec Room Simulator was used extensively by engineer Tom Jung during music recording for the film *Cotton Club*, to create room ambience that accurately matched the sound of archive recordings made at the actual Cotton Club and other speakeasies featured in the movie —*Editor*.]

The room program will determine the character of reverb in the Freeze mode. Another nice function is that when Freeze is turned off, the decay is that of the original program. One should follow entry instructions very closely (which is very easy), or some nasty sounds can occur. (For example, you must not chop off part of your input sample by pushing the enter button too soon, or the sharp edge of the "chop" will remain and reverberate in the room.)

Enhance mode is interesting in that it simulates a room without reverb, only pure reflection. As each reflection passes by the imaginary "microphone," it is cancelled from the program so that you only hear each original reflection; I would have to relate the feeling to that of tightly gated reverb. Room size affects this program, and the manufacturer recommends Enhance for the following applications: spatial spreading of a mono signal; synthetic binaural effects in headphones; and compression and intensification of sounds — I'm sure the user can come up with a number of



QRS SYSTEM BLOCK DIAGRAM

other creative uses.

All QRS parameters are adjustable by means of pushbuttons mounted on the well laid out front panel. Buttons market "+" or "-" move the parameters one setting at a time; alternatively, once activated by hitting one of these buttons a central rotary knob moves rapidly through the settings. When a button is pushed, the LED display flashes to show that the corresponding parameter is being adjusted.

All front-panel controls can also be accessed via an infra-red remote unit that is about a third larger than a cigarette pack — it's very handy not being tied to a cord. The unit itself can be stored back in a cool place with the IR sensor mounted anywhere in the room. Multiple remote controls can also be ganged together, and a front-panel display remoted to the console for easy view of operating parameters.

QRS is capable of storing 64 programs in non-volatile internal memory: There are eight files with eight locations in each file. Programs may be transferred to any location within a file, or even to other files for sensible organization. A safety circuit locks out and protects all but the first location of each file, which is where all parameters are adjusted in scratch-pad format. To store a setting, simply transfer a basic program to Location #1, Update, and Relocate to a protected location.

The Room Simulator is supplied with 14 factory preset programs from Quantec, and another 21 from Mar-

SUMMARY OF QUANTEC ROOM SIMULATOR SPECIFICATIONS

Reverberation program:
Room sizes: 1 to 10^6 cubic meters with 7 steps.
Decay time: 0.1 sec to 100 sec (up to 400 sec at 40 Hz).
Decay time at low frequencies: Coefficient of 0.1 to 10, with 11 steps in relation to selected decay time.
Decay time at high frequencies: Coefficient of 0.1 to 2.5 with 8 steps related to selected decay time.
Reverberation density: More than 10,000 per sec, depending on room size.
Density of resonance: Average of three per Hz of band width depending on room size.
Reverb: Pre-reverb delay 1 to 200 ms in steps of 1 ms, level -30 dB to 0 dB in steps of 1 dB; "OFF" Function
First Reflection: 1 to 200 ms in steps of 1 ms, level -30 dB to 0dB; "OFF" Function.
Enhance program: Simulation of rooms without perceptible reverberation.
Freeze program: Special loop program with infinite decay time to add any number of acoustical entries.
A/D converter code: 16 bit; Sampling rate: 20 kHz; Distortion: typical 0.05%.
Processor: 26 bit; Clock frequency 20.48 MHz.
Memory: approx. 2 Megabit of RAM.
Inputs: two, balanced, isolated by digital optocouplers; input impedance 13.2 kohms balanced, 6.8 kohms unbalanced; level adjustable -20 dBu to +6 dBu; headroom 12 dB above nominal level; RF-filter 18 dB/octave beyond 100 kHz.
Outputs: four, balanced; outputs 1 and 2 reverb plus first reflection; outputs 3 and 4 for quadrophonic use; output impedance 100 ohms balanced, 50 ohms unbalanced; minimum load 1 kohm; nominal level adjustable -6 dBu to +6 dBu.
Dynamics: better than 88 dB unweighted, typical 94 dB CCIR (valid for all decay times).
Frequency response: 20 Hz - 8 kHz, +1/-3 dB.
Connectors: XLR-3.
Dimensions: Standard 19-inch width; 2U height; 260 mm depth.
Weight: 5.5 kg.
Price: QRS $9,995; IR Remote $750; QC-3 Software $399.
US Distributor: Marshall Electronic, 1205 York Road, Suite 14, Lutherville, MD 21093. (301) 484-2220.

shall, including an "EMT plate" that sounds better than a lot of plates I've used. (Each preset can be fully adjusted from its factory value.) Having done a multitude of concert gigs across the country, it was fun listening to the accuracy of the unit's full concert hall, compared to its empty concert hall. QRS has several different rooms, plus some different settings — such as a 1,000-litre "Oil Drum" — and my favorite for kick drum, the "Submarine." These programs provide the user with a good starting point from which to create their own environments.

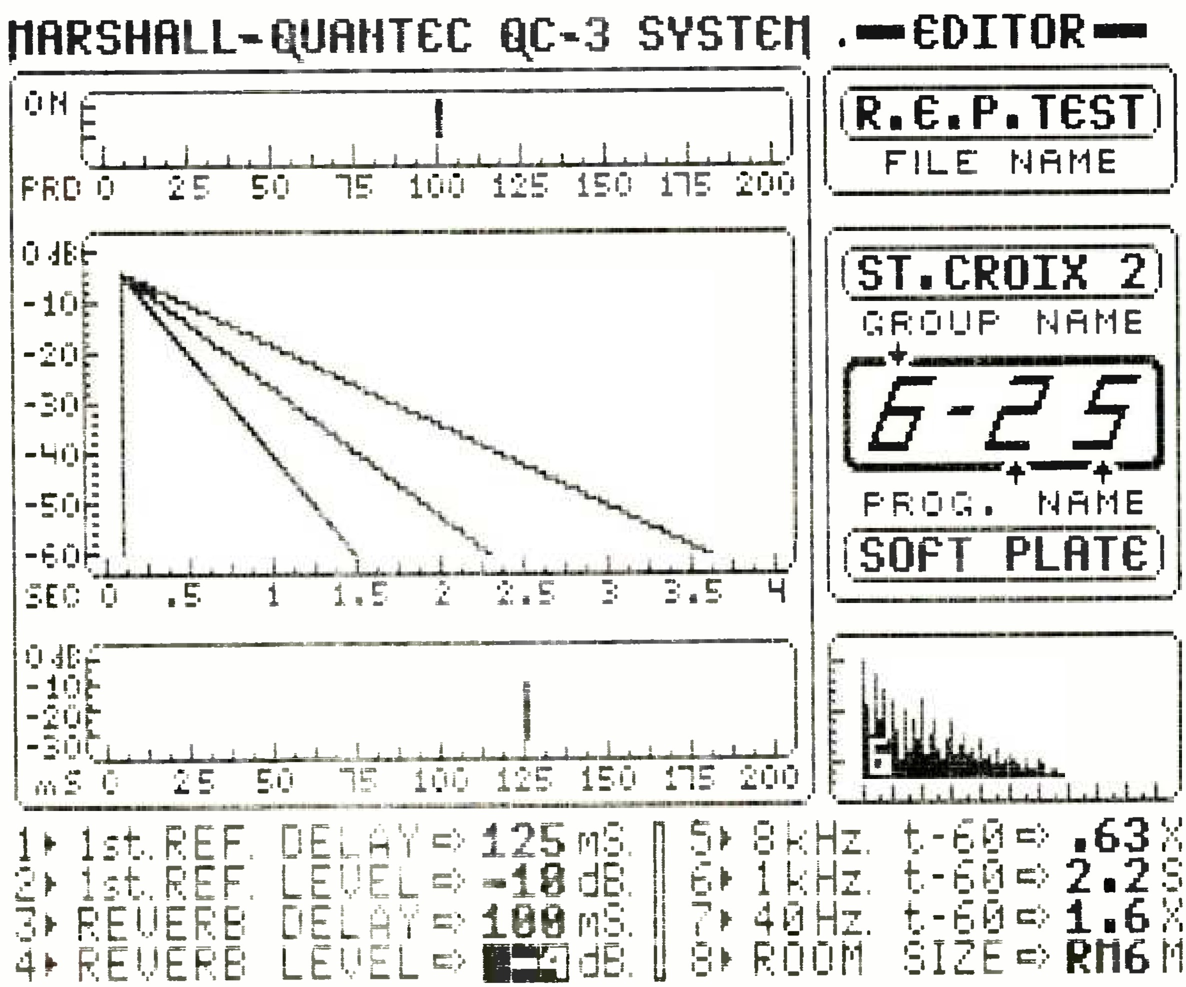
QRS has truely realistic and smooth sounding reverb, part of which is due to the complexity and sheer number-handling capability of the processor. I was able to input much higher levels with more transients into QRS than any other digital reverb I've used, yet with none of the digital overload normally experienced. Also, the input signal is processed separately for the left and right channels, and the signals mixed in the reverberation field, thereby providing a more realistic stereo and binaural effect.

QRS Infra-Red Remote Control provides full functions in a hand-held unit

Beyond normal use in the recording and production studio, applications of the QRS is film dubbing and ADR are enormous. Highly realistic environments for voice and sound effects can be achieved, and set up quickly with a little knowledge of the properties of reverb. (Of course, the same applies to music recording for the creation of really powerful reverb effects.)

Just released by Marshall Electronic is the QC-3 Software Control Program that operates on Apple IIc, IIe and II+ computers. The computer connects to the QRS through the remote-control interface, and can store up to 1,000 control settings. The Apple's VDU graphically and numerically displays RT_{60}, pre-delay, initial reflection, dispersion density, room size, program name in alpha-numerics, and all front-panel parameters, as shown on the accompanying illustration. In addition to all of this, the software generates a code which, once recorded on tape, will automatically change the QRS programs at specified cues, a feature that has numerous applications in music, film mixing, and theatrical performance. ■■■

Typical Screen Display from the QC-3 Software Control Program developed by Marshall Electronics to enable an Apple IIc/IIe/II+ to control the QRS via an RS-232C connection.



New Products

PROGRAMMABLE DIGITAL METRONOME/SYNCHRONIZER FROM AXE

The KT-1000 is a microprocessor-based, Programmable Digital Metronome Synchronizer that will store 50 different user-programmable tempos. Various tempos can be programmed to run for a specific number of beats; automatically start each tempo from SMPTE coded videotape, film, or audio tape; and not only put out a metronome pulse for live recording, but automatically control MIDI-equipped devices, including accleratons and retards, to ensure that everything runs in perfect synchronization.

Tempos can be entered in either beats per minute or frames per beat. There are four frame formats: 24 (film), 25 (EBU), 29.97 (NTSC Drop Frame), and 30 NTSC. The duration of the tempo can be entered in either the total number of beats or a specific length of time, accurate to 10 milliseconds (e.g.: 16 beats or 17.34 seconds). The KT-1000 can be programmed to go from one tempo to another automatically, accelerating and retarding as needed within a specific number of beats. Tempos can be started automatically from SMPTE timecode off either video tape, audio tape, or SMPTE-coded film.

Besides putting out its headphone click, the unit, via a MIDI interface, outputs the required number of pulses to automatically control synthesizers, sequencers, and drum machines.

The KT-1000 will also save and load the user's program to tape, or to a non-volatile Memory Card. (In the first quarter of 1985 an add-on RS-232C interface buss will be available so that the user's program can be derived from a computer, or sent to either a VDU or printer.)

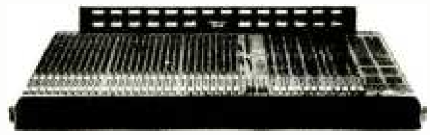
ARTISTS X-PONENT ENGINEERING

For additional information circle #98

SOUNDTRACS MICROPROCESSOR-CONTROLLED CM4400 CONSOLE

A single-button routing system permits all input and output modules to be identical in "hardware" construction. The result is that there are only two kinds of modules — input and output — that are the same regardless of the number of busses; only the software changes.

An RS232 port has been added to allow the CM 4400 to communicate with any PC which has an anynchronous input. By writing software that allows the track sheet to be preprogrammed, each memory position can be shown on the terminal as functions, not numbers. Software updates are now being explored, the company says, to enable the mixer RAM to be externally steered by a SMPTE clock or track.



The CM 4400 is transformerless, and all inputs are electronically balanced. (Output transformers, as well as P&G faders, are optional.) The board uses 5532, 5534, and TLO series chips, and discrete mike pre-amp with IC buffer. The line output features a screwdriver-adjustable gain control allowing for any type of tape machine input +4 dB or -10 dB level. The slew rate is a quoted 10V per microsecond, eliminating group delays and adding to improved sonic quality.

SOUNDTRACS, INC. USA

For additional information circle #99

SHURE LOW-PROFILE SM91 BOUNDARY MIKE

The SM91 condenser microphone is a low-profile model designed for surface-mounted applications where a unidirectional pickup pattern is desirable, and takes advantage of the well-known principle of boundary effect. Because of this principle, placing an SM91 sufficiently close to a barrier or boundary will cause it to perform with as much as 6 dB higher sensitivity and approximately 3 dB greater rejection of random background noise.

"We're presenting the unidirectional SM91 as a viable alternative to the omnidirectional 'pressure zone' microphones currently in use," says John F. Phelan, Shure's marketing manager, professional products. "Since this is the first unidirectional microphone that utilizes boundary effect, it has many advantages over other 'pressure zone' models on the market."

According to Phelan, these advantages include minimized low-frequency noise and rumble, less tendency toward feedback, and avoidance of phase cancellation. In addition, the half-cardiod pickup pattern of a surface-mounted SM91 permits the microphone to operate with much less reverberation and "muddiness" than omnidirectional surface-mounted models. The unidirectional pattern also allows for effective isolation without the need for physical isolation barriers often used with "pressure zone" models.

At the heart of the mike is a new cartridge that is said to provide high output plus a wide, flat frequency response for accurate sound reproduction and excellent off-axis performance. A separate pre-amplifier may be powered either by two standard 9-volt batteries or by an 11 to 52 VDC phantom supply, and includes a 12 dB per octave LF cutoff switch. The SM91's microphone base is constructed of rugged, matte black enameled metal.

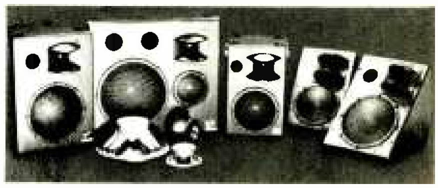
User net price for the SM91 is $300.

SHURE BROTHERS, INC.

For additional information circle #100

FENDER 2800 SERIES LOUDSPEAKER SYSTEMS

The five new models include three systems for general use, plus two models configured as on-stage floor monitors. All systems use high-power cone drivers with special constant-directivity HF horns and 18 dB per octave crossover networks to achieve "smooth, wide-range response and high output capability."



Model 2821 is a three-way design including 18-inch woofer and eight-inch midrange drivers. With a 200-watt continuous music power rating (and 100 dB per watt per meter sensitivity), the 2821 is recommended for high-level indoor and outdoor applications. Like the smaller 2841 two-way system (with 15-inch woofer), the 2821 is said to offer an extremely rigid trapezoidal "wedge" enclosure that stacks efficiently into splayed arrays with any desired horizontal coverage angle.

For smaller indoor installations, the 150-watt Model 2851 features a 12-inch woofer and constant-directivity 90- by 40-degree horn. Both it and the 2841 are also available in stage monitor versions (numbered 2842 and 2822), housed in asymmetrical enclosures that provide a choice of two different tiltback angles.

FENDER MUSICAL INSTRUMENTS

For additional information circle #101

NEW DYNAFEX NOISE-REDUCTION SYSTEMS FROM MICMIX

The new Model DX-2 stereo Dynafex and DP-1 mono Dynafex system both incorporate a brilliance control.

The DX-2 can be operated as a stereo device, or to provide two independent channels. Each channel includes a Threshold control that is adjustable from 0 to -30 dB; this control determines the level at which downward expansion begins to occur. Also included on each channel is a continuously adjustable Brilliance control that allows the

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READING
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REMEMBERS START AND CUES

UNIVERSAL MODULAR SYSTEM

SOLVES ANY SYNC PROBLEM NOW AND IN THE FUTURE

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NEW SOFTWARE RELEASE AUGUST 1984

SMPTE-CODE generator/reader switchable for 24F, 25F(EBU), 30 non-drop and 30 drop-frame-code
32 CUE-POINTS, autocorrection to the metronome, improved CUE SETTING, CLEAR CUE, CLEAR ALL
8 free programmable sequences for TEMPO CHANGES, 32 steps each, presets
DATA TO TAPE via the SMPTE chanal saves START, CUES, TEMPOS

NEW INPUT MODULE

reads any* clock or FSK from metronome up to 1536 clicks, multiplies and divides
reads natural DRUM TRACKS, manual trigger, sound/trigger conversion
LOOP to pass dropouts, missing drumbeats or even breaks
PUNCH-IN funktion even when working without SMPTE

MIDI SYSTEM CLOCK

* AMDEK, DRUMULATOR, FAIRLIGHT, KORG, LINN, MICROCOMPOSER, MOOG, MXR, OBERHEIM, PPG WAVE, ROLAND, SEQUENTIAL CIRCUITS, SIMMONS, SYNCLAVIER II - - - AND ALL OTHERS

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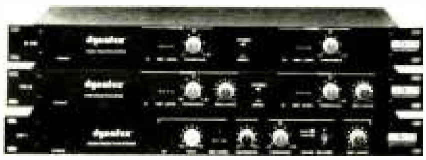
New Products

user to replace some of the HF compenents that may be aurally perceived as lost during the noise-reduction process. The manufacturer claims that by utilizing this control, the user can end up with a signal that is free of unwanted noise, while maintaining the original brightness or crispness in the HF signal content. The DX-2 also includes a reference level switch on each channel that allows the unit to be operated at -10, 0, +4, or +8 dB levels.



The single-channel DP-1 also incorporates similar controls. In addition, this device includes a switch that allows the user to extend the Threshold control range by 20 dB; the user may also select between a fast, medium, or slow release time for the downward expander circuitry. The DP-1 includes a gain control that prvovides up to 40 dB of gain, allowing easy interface with signal levels below typical reference levels. A Bandwidth control, adjustable from 3 kHz to over 20 kHz, is also provided that allows the user to determine the overall bandwidth that the filter operates.

MICMIX AUDIO PRODUCTS, INC.

For additional information circle #103

SOLID STATE LOGIC UNVEILS SL5000M SERIES PRODUCTION CONSOLE

The new console system incorporates many features specifically designed to handle production requirements of the new Multichannel Television Sound transmission standard. In addition to the main stereo and mono program outputs, the SL5000 can be specified with up to four independent stereo outputs, and eight stereo audio subgroups, which may be configured to handle any combination of international sound split feeds, mix minuses, and stereo plus SAP mixes.

Mainframes for the new system will accept from eight to 56 mono or stereo inputs. Special panning and width enhancement circuitry is provided to simplify the matching of stereo perspective with video. Up to six stereo or 12 mono auxiliary sends may be provided, along with as many as eight stereo IFB/Cue select matrices. (These features are said to speed and simplify the creation of the multiple cue and communication feeds required for stereo teleproduction.)

Combining the latest developments in thin- and thick-film technology, SSL has developed a new series of hybrid chips that replace the earlier "op-amp plus components" type of subassembly, and which substantially reduce the size, weight, complexity and power consumption of the system's basic building blocks. For example, a single SSL Hybrid Chip, providing a totally balanced and ground-free output capable of driving a 600-ohm load at +28 dBm, takes up an area of about 40 × 15 × 5mm.

The mainframes can be visualized as having a number of "positions," which are the vertical channels running from the fader through to the penthouse. Each position is 40mm wide and divided into a number of "rows," which are the horizontal slots running across the mainframe. Each row is 150mm high, with the exception of the penthouse row, which is slightly higher. Thirty-six standard mainframes are available. There are three basic mainframe models, providing either four, five or six rows of horizontal busses; each model is available in two styles, providing either eight or 16 Master Positions. All models and styles are available in frame sizes accepting up between 16 and 56 mono or stereo channels.

An initial family of 28 Eurocard audio and control "cassettes" have been developed. The miniaturization afforded by SSL Hybrid Technology has made it possible to include complete control and data circuitry, along with specialized audio electronics, in each of the 40 × 150mm cassettes. All switching within the console is totally electronic, and the vertical and horizontal bus arrangement provides address lines to all cassettes, regardless of their individual positions. This capability enables extensive master control facilities to be incorporated within an essentially custom environment. (For example, Master Controls on the Auxiliary Output cassette allow the engineer to switch all local sends to each auxiliary bus on or off [and pre or post] at the touch of a single button.)

The SL5000M will be available with two levels of computer assistance. The first level, SSL Instant Reset, will store all switch settings, allowing broadcasters to instantly reset the console between any number of master and local configurations. All channel cassettes are also addressable by the SSL Total Recall system, allowing the exact values of all variable controls to be stored and recalled with a quoted control accuracy of 0.25 dB. The SSL Studio Computer also interfaces with the new console series, providing complete dynamic mixing automation with integral synchronizer and machine control.

SOLID STATE LOGIC, INC.

For additional information circle #104

OTARI INTRODUCES NEW MTR-20 SERIES TRANSPORTS

The MTR-20s will be available in ¼-inch two-track (with and without center-track timecode), half-inch two- and four-track formats, and feature microprocessor-based automatic alignments of record level, Hi EQ, Mid-high EQ, bias, and phase compensation. Four speeds are available — 3¾, 7½, 15 and 30 ips — and the machine will accommodate reel sizes up to 14-inch diameter.

The transport has been designed specifically for easy control by SMPTE/EBU timecode-based editors, machine controllers or synchronizers. In addition to the parallel remote connector, optional serial communications ports are available. According to Otari, the EPROM-based protocol of these ports will allow simple, rapid updating to follow evolving standards for intelligent machine control.

Other features include: extensive user programmable functions for both transport and electronics; an integral four-point search-to-cue, RTZ and Search-start, ±45% varispeed; direct-coupled active balanced I/0; cue speaker and headphone amp; tape time/speed display; a cue shuttle level, and a controlled wind "library pack" mode.

Pricing is as follows: MTR-20-C (¼-inch, two-channel) $11,000; MTR-20-CT (as above with timecode channel) $12,500; MTR-20-H (half-inch, two-track $12,000; and MTR-20-Q (half-inch, four-track) $13,000.

OTARI CORPORATION

For additional information circle #105

AUDIO INTERVISUAL DESIGN INTRODUCES TWO NEW TIMECODE PRODUCTS

"The Stripper" is a battery powered unit that reshapes SMPTE timecode and field rate sync (59.94 or 60 Hz sinewave). The unit has one timecode input, and separate outputs for re-shaped timecode and sync waveform.

Typical applications include reshaping timecode for dubbing or for use with marginal readers; conditioning waveforms or pulses for synthesizers and drum machines, etc.; stripping field rate sync for film synchronizing; deriving 60 Hz pulse rate from 24-frame timecode; locking a 30-frame generator to 24-frame timecode; and using a time-coded tape as an emergency code source.

SMPTE Timecode input range is -20 to +20 dBm, differential, bridging, while the timecode output is slew limited, level adjustable balanced +4 dBm, and the field rate sync (48 to 70 Hz) is balanced +4 dBm.

"The Driver" is a resolver that accepts SMPTE timecode input and delivers Quadrature Pulses at selectable rates to drive all popular 16mm and 35mm film projectors and dubbers, allowing a film chain to be resolved to timecode without expensive, mechanical interlock systems.

Timecode input range is -20 to +20 dBM, differential, bridging, while the outputs comprise Quadrature Pulses at selectable rates (for example, 2.4 kHz with Ramp up and Ramp down times adjustable from 0.5 to 5 seconds; forward direction only), and a 60 Hz sync pulse.

Pro-user price of The Stripper is $249, and The Driver $349. (FOB Los Angeles).

AUDIO INTERVISUAL DESIGN

For additional information circle #106

GATEX FOUR-CHANNEL NOISE GATE/EXPANDER FROM U.S. AUDIO

At the heart of Gatex is the new Valley People TA-104 voltage controlled amplifier.

By virtue of its distortion-free operation and wide dynamic range, the TA-104 VCA is said to allow the unit to process audio signals without coloration.

Feed-forward control circuitry provides accurate gain control without instability caused by control "lag" commonly found in less expensive processors, while DC control of all functions eliminates "noisy pot" problems because no audio signals pass through the front panel controls.

The unit's variable threshold encompasses the range of levels from -40 to +20 dB, providing the versatility to process all types of program material. A complementary range control enables the user to adjust the amount of maximum attenuation from subtle noise reduction 80 dB cut-off. A Program Controlled Sustain automatically lengthens the release time as dictated by program content. As a result, desirably short release times may be employed without creation of unwanted distortion.

The Gatex mode select switch permits the unit to perform "hard" noise gating, 1:2 expansion, or unobtrusive noise reduction. In all modes, "turn on" noise is eliminated by means of Program Controlled Attack, which alters attack time according to the demands of the material being processed.

"Keying" also is made possible via the Gatex source switch.

Recommended user price of the Gatex is $399.

U.S. AUDIO, INC.

For additional information circle #107

SOUNDCRAFT SA SERIES OF POWER AMPLIFIERS

The new amplifier series is said to eliminate the concept of TIM distortion from circuitry; to combine MOSFET and bipolar designs to allow the amplifiers to respond dynamically to program as through they were much larger in average power; to offer very high current and voltage slew rates, and extremely low phase shift. All the amplifiers are fully protected and compact for smaller than usual rack requirements.

The full line consists of three models: the SA2000 rated at 435 watts RMS per channel into 8 ohms (3,000 watts RMS per channel for 5 mS); the SA600 rated at 150 watts RMS per channel into 8 ohms (700 watts RMS per channel for 5 mS); and the SA150 rated at 85 watts RMS per channel into 8 ohms (450 watts RMS for 5 mS). Prices are $749, $949, and $1,975, respectively.

SOUNDCRAFT ELECTRONICS, INC.

For additional information circle #108

For additional information circle #109

New Products

CONNECTRONICS ANNOUNCES NEW RANGE OF SECK MIXERS

The SECK 62 and the SECK 122 offer six- or 12-input channels with two outputs in a low-profile, portable package: the total unit, which is fully metal encased, is only two inches deep. The rugged design is enchanced by the use of a double-sided, fiberglass PCB and by the elimination of any wired connections; all input and output connectors are mounted directly onto the circuit board, resulting in higher reliability than many conventional mixers, the company claims.

The microphone/line input switch on each input channel selects either a low or high impedance input. Both are electronically balanced, and together cover the range -55 to +10 dBm, with 25 dBm overload margin. Three-band EQ offers ±15 dB of midrange sweep control from 330 Hz to 6.5 kHz.

Each channel has two pre-fade auxiliary sends for monitoring, foldback, or special effects, and two post-fade sends to feed effects or for specialized house PA or recording. In addition, each channel has a pre-EQ insert point to cater for extra limiting, delay, etc., on any individual input. Solo switches are provided on all inputs, four master auxiliary sends and auxiliary returns.

CONNECTRONICS CORPORATION

For additional information circle #112

CROWN INTRODUCES PCC-160 PHASE COHERENT CARDIOID MICROPHONE

The new PCC-160 is a surface-mounted supercardioid microphone intended for professional applications on stage floors, lecterns, conference tables, and news desks. When used as a "footlight" microphone for drama, musicals or opera, the PCC is said to provide louder, clearer sound pickup than previous microphones.

Similar to the PZM, the PCC is designed to be used on a relatively large boundary surface. Unlike the PZM, however, the Phase Coherent Cardioid uses a sub-miniature supercardioid mike capsule whose unidirectional polar pattern increases gain-before-feedback, reduces unwanted room noise and off-axis pickup.

Since the microphone capsule is placed on a boundary, direct and reflected sounds arrive at the diaphragm coherently, or in-phase. The benefits are described as a wide, smooth frequency response free of phase interference, excellent clarity and reach, and a "half-supercardioid" pattern (based on the hemisphere created by the large boundary plane).

Capable of withstanding up to 120 dB SPL without distorting, the electret condenser capsule provides a quoted frequency response from 50 Hz to 18 kHz; sensitivity is -52 dB re: 1 volt per microbar and self-noise less than 22 dBA. Output impedance is 150 ohms, balanced.

Suggested list price of the PCC-160 is $249.

CROWN INTERNATIONAL

For additional information circle #113

BARCUS-BERRY ELECTRONICS UNVEILS BBE 202 DIFFERENTIAL LOAD REACTANCE COMPENSATOR

The new BBE 202 is a multiband, program-controlled signal processor that can be employed to improve the overall sonic clarity of virtually any reproduced sound, by utilizing high-speed dynamic gain-control circuitry to audibly improve the reproduction of program transients. Such processing is said to add "brightness" and "presence" without introducing the undesirable stridency so often characteristic of "equalized" sound, especially at peak levels. The unit also increases voice intelligibility by eliminating frequency-band masking when important sibilant and consonant elements are represented in the program signal.

The dual-channel unit is packaged in a 19-inch, rack-mountable chassis that occupies two standard EIA spaces and is 7 inches deep. Normal set-up requires only simple adjustment of a single control for each operating channel. This is preferably done while listening to typical program material, but can be accomplished without monitoring, if necessary, by sending program and utilizing a front-panel LED display which indicates the sensitivity threshhold of the processing circuitry. Once installed, all processing functions are fully automatic.

Phase adjustments are primarily directed toward preventing high-frequency time lag (transient distortion) and the automatic gain changes are based on interband program amplitude ratios. Swept frequency response of the system is said to be essentially flat from 20 Hz to 20 kHz in both the operating and electronically-buffered bypass modes. Amplitude changes are developed only in direct response to application of a spectrally-diverse program signal.

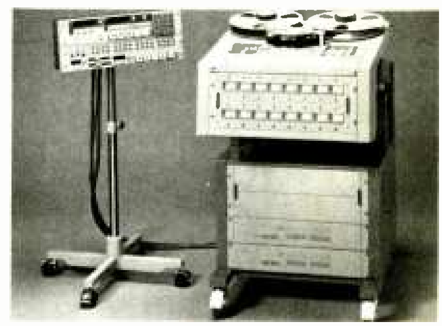
BARCUS-BERRY ELECTRONICS, INC.

For additional information circle #114

NEW MS-16 ONE-INCH 16-TRACK FROM TASCAM

Features and capabilities of the new MS-16 include a rear panel SMPTE connector with TTL logic lines for interfacing with most popular controllers, a transport said to be strong enough to handle SMPTE interlock or other unusually tough assignments, and a full three-motor servo system for positive tape tension control throughout quick lock-ups and stops. A console is available that allows positioning the meter panel in front or overhead the transport.

Hyperbolic head geometry with micro-radii means "head bumps" can be reduced to less than 1 dB, the company claims. There is no need for switching to repro head (except during alignment) since the record/sync head and the repro head feature identical performance.



Amplifiers are all direct-coupled for lowest distortion and optimum LF response. Improved transient response and phase characteristics are obtained with the MS-16's first-stage sync and replay head amplifiers, which use differential paired, ultra-low-noise FETs. The multitrack also has separate low-frequency compensation alignment for record/sync and repro heads, plus both +4 dBm balanced XLR and -10 dBV unbalanced RCA outpus.

Optional accessories include a 10-point autolocator, and basic function remote control, CS65 console, and a dbx noise-reduction unit.

TASCAM PROFESSIONAL PRODUCTS

For additional information circle #115

MIDITRACK II DIGITAL MIDI RECORDER SOFTWARE FROM HYBRID ARTS

The new software provides a 16-track digital MIDI recorder, synchronizer, and MIDI remote control that uses the company's MidiMate to interface with an Atari 800XL personal computer, at an affordable cost.

Features include 16-track overdubbing, punch-in/punch-out, autolocate, full MIDI channel assigning, velocity encoding, pitch and mod-wheel recording, program change recording, transpose and quantizing, step editing, and a variety of sync in/out interfaces.

MidiTrack II is designed to take advantage of the full MIDI specifications and therefore works with all the different MIDI keyboards and drum machines, regardless of manufacturer.

It allows the user, in real time, to selectively turn tracks on, off or solo, change tempo, or re-assign MIDI channels. Auto locate and punch in/punch out are available at the touch of a button. All operations are non-destructive. With operations such as quantizing or transposing, the original track is preserved while the result is saved on another data track. Full MIDI mode commands are supported, such as local on/off or poly/mono modes.

The program screen shows all statuses, as well as note on/off display. Sequences can be 6,500 events long — enough for a five- or six-minute song with all parts. Three entire recordings can be saved per disk. Two forms of metronome are offered — visual and audio — in addition to a complete range of synchronization features; MidiTrack II can operate in slave or master sync mode. The internal clock has a range 2 to 750 beats per minute, while the external sync input can come from a drum machine or a tape deck. The unit will output TTL and MIDI clock.

MidiTrack II is priced at $349, and includes a software disk, MidiMate interface, two MIDI cables and a User's Guide.

HYBRID ARTS

For additional information circle #116

FM ACOUSTICS MODEL FM 236 ELECTRONIC CROSSOVER

Their FM 236 features fully discrete class-A circuitry throughout, proprietary linear-phase filters that are said to achieve 36 dB per octave attenuation without any overshoot, and perfect step response.

The unit's guaranteed specifications include a minimum of 75 dB CMRR, crossover frequency accuracy within 2% of rated frequency, rise time of 500 nanoseconds, and channel separation better than 70 dB at 10 kHz.

FM ACOUSTICS, USA

For additional information circle #117

ROCKTRON INTRODUCES MODEL 300 COMPRESSOR

With built-in HUSH II noise reduction, the Model 300 is described as the only compressor/limiter to feature dynamic filtering and low-level expansion, solving the problems encountered in many compressor/limiter applications. In applications where maximum gain reduction is required, other compressors fall short, the company claims, suffering from an increase and modulation of the noise floor.

The unit simultaneously performs the functions of compression, peak limiting, low level expansion, and dynamic filtering.

The Model 300 provides program-dependent logarithmic compression, offering the "smoothest transition into compression." The compression adjustment simultaneously controls the threshold of compression and input level, to maintain a constant output level independent of the amount of gain reduction used. When the input signal level exceeds the threshold, gain reduction begins and logarithmically increases in relation to the input signal level.

It also offers frequency-dependent limiting/compression: by inserting an outboard equalizer into the side-chain input, the Model 300 becomes a de-esser, and can also be used as a pre-emphasis compensated limiter.

Suggested retail of the Model 300 is $390.

ROCKTRON CORPORATION

For additional information circle #118

For additional information circle #119

New Products

RAMSA UNVEILS TWO NEW CONSOLES RANGES

The WR-T812 and WR-T820 recording consoles feature 12 and 20 inputs, respectively; the -T812 is specifically tailored for eight-track work while the -T820 offers eight- and 16-track capability. Both boards feature the ability to simultaneously mix incoming signals with tape playback signals during overdubbing, without the need to repatch.

Each input offers pushbutton selection of tape signals and electronically balanced mike and line. Phantom 48-volt power is individually switchable for each channel, and a phase-reversal switch is included for line and mike inputs. Each input channel has its own direct output and insertion point for external signal processors. A sweepable three-band EQ section with an on/off switch and an 80 Hz high-pass filter is also included.

Two different metering configurations are available: the "basic" metering configuration consists of 10 LED bar graphs that monitor all group signal, master L/R, auxiliary sends and stereo solo metering; the second configuration comprises a meter bridge. The WR-TR812 meter bridge houses eight bar graphs for input channels one through eight (for eight-track mixdown) and 10 VU meters for output group signals, master L/R, auxiliary sends and stereo solo; the WR-T820 bridge features 16 LED graphs and 10 VU meters.

Suggested retail price for the WR-T820 is $4,995 and, for the WR-T812, $3,995.

• The WR-S Series of stereo mixers consist of the eight-input WR-S208, 12-input WR-S212, and 16-input WR-S216. All three models can be used with a variety of input sources, including tape decks, CD players, turntables, VCRs, and direct line inputs. Two channels have stereo input for both line and phono inputs. All mono inputs are electronically balanced for mike and line inputs.

Three send circuits are provided: a pre-fader foldback circuit; a post-fader effect send; and a switchable pre-/post-fader send for either foldback monitoring or effects. The switchable pre-/post-fader send can accommodate two different monitor or effects sends, or a combination of the two independently.

Three main outputs are provided: A, B and main mono output; the level of output A or B can be adjusted independently of the summed output. (Sub-outputs A and B might be used for stereo recording, while the main output —A and B summed — provides monitoring to feed a PA system.)

The Model WR-S208 has a suggested retail price of $1,295, the WR-S212 $1,695, and the WR-S216 $1,995.

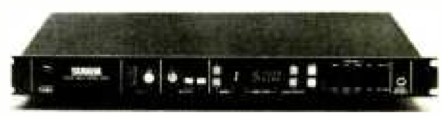
PANASONIC RAMSA

For additional information circle #120

YAMAHA MODEL D1500 DIGITAL DELAY

The D1500 features 16 programmable preset memories, and a zero to 1.023-second delay range selectable in one millisecond increments. Quoted frequency response is 20 Hz to 18 kHz.

The 16 memory banks (0-9, A-F) can be programmed with different delay settings, with programs A thru F preloaded at the factory. A program can be recalled in one of three ways: using front-panel controls; a foot switch; or remotely from a MIDI keyboard. The MIDI programming capability allows each synthesizer voice (up to 128 synthesizer presets) to automatically select one of the D1500 programs, or bypass.



Ten pushbuttons assign the settings for each program: time; feedback lowpass filter; feedback level (for regeneration); invert (inverts the phase of the delayed signal); mix; rate, wave, and depth (for the LF oscillator); and two switches to control the MIDI channel recall. Each of these parameters is programmed using the two data entry switches. An LED display shows the memory bank selected and the delay time or the parameter of the program being programmed.

Suggested retail price of the D1500 is $895.

YAMAHA INTERNATIONAL CORPORATION

For additional information circle #121

NEW JRF CONVERSION RETROFIT HEAD ASSEMBLY FOR AMPEX ATR SERIES

The newly introduced half-inch/two-track ATR assembly is said to offer easily accessible adjustments for azimuth and head wrap that are extremely "smooth" to ensure optimum performance. The assembly also includes premium quality, long wearing SAKI magnetic heads for added performance and reliability.

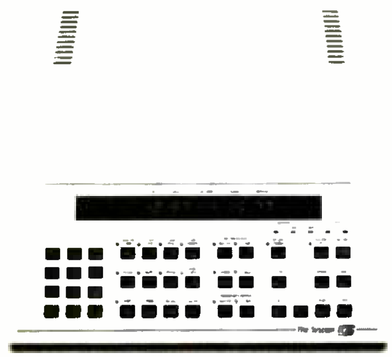
The company has also developed conversion packages for MCI JH-110A and -110B, Ampex AG-440 and 3M analog machines.

JRF MAGNETIC SCIENCES

For additional information circle #122

"THE SYSTEM" SMPTE CONTROLLER/SYNCHRONIZER FROM BTX

The System is a dual-transport SMPTE controller/synchronizer that requires only an 18- by 18-inch space, stands five-inches tall and weighs just 20 pounds. The unit's built-in keyboard features 40 dedicated function keys and a 10-digit LED time code display that facilitates autolocation and editing.



Features include pre-programmed loops with optional preview; pre-programmed or "hot" Master or Slave record-in and -out; offsets up to 24-hours; selectable interlock speed and type; synchronization accuracy within 333 microseconds; a system memory; plus standard transport control and autolocation capabilities.

According to Michael Padovano, Director of Marketing at BTX, "End-users were excited at the prospect of having so many capabilities in a low-cost unit the size of a small personal computer." He added that BTX believes The System to be the most compact full-function controller/synchronizer on the market today.

THE BTX CORPORATION

For additional information circle #124

AUDIX ANNOUNCES UD-260 DYNAMIC MICROPHONE

The high-output, low-impedance mike incorporates a new air-suspension design and an integrated capsule system for easy field replacement. Designed as a rugged, high-end vocal microphone for stage and live applications where an on-off switch (lockable) is desired, the UD-260 is said to provide a smooth response from 50 Hz to 18 kHz without harsh midrange peaks, and a tight cardiod pick-up pattern for higher gain before feedback. It is available in black, non-reflective matte gray and six colors with matching cables.

Suggest list price of the UD-260 is $179.

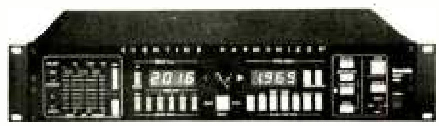
AUDIX CORP. OF AMERICA

For additional information circle #125

EVENTIDE INTRODUCES NEW H969 PROPITCH HARMONIZER

The new model employs a newly designed digital intelligent splicing algorithm system — dubbed ProPitch — to deliver cleaner pitch-change performance without glitching over a wider frequency range than ever before. The company has also used 16-bit PCM linear coding for the first time in the H969 Harmonizer unit.

New features include a dozen pitch-change presets, enabling the user instantly to set a precise minor third, major third, fifth, seventh, or octave of pitch change; each can be selected as a sharp or flat. In addition, separate coarse and fine adjust controls enable the user to easily set precise pitch ratios.



Full bandwidth delay has been increased to 1.5 seconds, with a further increase to three seconds at half bandwidth. The user can choose and save any delay times for instant recall. The full delay range is also available in repeat and reverse modes. Doppler effects have been added to the H969, and flanging is also available.

EVENTIDE, INC.

For additional information circle #126

LINN 900 COMBINED MIDI KEYBOARD RECORDER AND DIGITAL DRUM UNIT

Described as the first product to integrate a MIDI-compatible keyboard recorder and digital drum machine in one unit, with programming parameters identical for both, the new Linn 9000 Keyboard Recorder (also known as a "sequencer") memorizes every aspect of performance — dynamics, pitch bends, modulation and synth patches — simultaneously for as many as 16 MIDI-equipped polyphonic synthesizers (with a maximum of 32 tracks.)

The unit is said to embody all current technology for such devices (including the LinnDrum), while introducing many exclusive features, including velocity-sensitive front-panel keypads (or rear-panel inputs for electronic drum pads); programmable hi-hat decay that permits highly accurate simulation of drummers' variable foot pressures; built-in mixer with separate faders assigned to each sound for selective memorization of volume, pan and tuning; "Repeat" function to provide quick programming of rolls, constant 16th notes, etc.; versatile tempo programming via count-off "tap" or numeric entry (including

New Products

tenths of a beat); plus 18 digitized drum and percussion sounds (two crash and two ride cymbals, four toms, two congas, bass, snare, hi-hat, sidestick, cowbell, tambourine, cabasa and handclaps.)

Recording and editing functions have been designed to simulate the familiar operation of a multitrack, with record, play, fast foward, and rewind buttons.

Retrofittable options include a plug-in audio sample board; a 3½-inch disk drive to augment present cassette capability for loading or storing drum and synth programs, as well as drum sound samples; and an additional circuit board to implement SMPTE interlock.

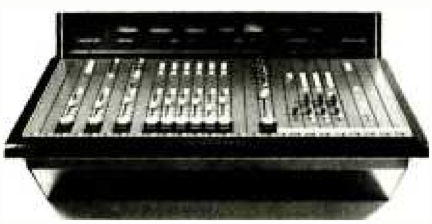
The Linn 9000 carries a suggested retail price of $4,990.

LINN ELECTRONICS, INC.

For additional information circle #128

SOUNDCRAFT ANNOUNCES SAC 2000 STEREO ON-AIR CONSOLE

Since the new SAC 2000 features a built-in cartridge machine sequencer, cart machines do not require external strapping or sequencing devices. The console also features remote activation of input modules or cart sequences from separate studios, or from remote broadcast sites via a special sub-audible tone unit to allow spot or music pre-sets at the console to be activated from the field.



Other features include three stereo plus mono outputs; full metering; multiple input selection with logic-follow; four-channel telephone mix-minus; multiband equalization; delay control; and universal machine-control logic.

SOUNDCRAFT BROADCAST DIVISION

For additional information circle #129

NEW MX-70 ONE-INCH MULTITRACK FROM OTARI

Like the MTR-90 multitrack, the new MX-70 series of eight- and 16-track on one-inch transports features a microprocessor-governed constant-tension, servo-controlled transport: high-quality audio electronics with timed bias ramping for gapless insert recording at any speed; logic-interlocked controls; a remote controller and interface connectors for any SMPTE/EBU timecode-based video editing system, machine controller or synchronizer. An optional conversion kit allows operation with half-inch eight-track tapes. Additionally, an RS-232C or RS-422 serial communications port may be ordered.

A full-function Remote Session Controller comes standard with the machines, and an optional autolocator with multiple memory storage, search and repeat shuttle capabilities are available.

Selected features include field-convertible speed pairs (7½/15 ips or 15/30 ips); adjustable phase compensation; switch selectable IEC/NAB equalization; switching logic for interface to any noise reduction system; transformerless, active, balanced in/out (+4 or -10 dB) and an LED multi-function tape/time display.

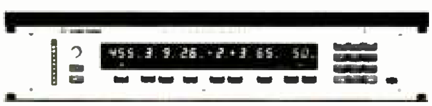
Pricing is as follows: MX-70 8-track $12,500; MX-70 8/16 convertible $13,500; and MX-70 16-track $14,950.

OTARI CORPORATION

For additional information circle #130

KLARK-TEKNIK DN780 REVERBERATOR/PROCESSOR

The new unit is said to incorporate entirely new processor algorithms based on a mathematical theory never before applied to this type of equipment. These new algorithms reflect the need to process a high number of reflections to realistically simulate a particular environment, and the internal architecture of the DN780, using 16-bit linear A/D and D/A converters, and a 32-bit arithmetic processor, provides the necessary computing power to handle these reflections. This results in what the company refers to as Added Density™ reverberation.



The DN780 features LED displays of reverb parameters, and "nudge" controls for varying any of the operational settings. There are 20 factory preset Reverb Programs for Room, Hall, Chamber and Plate, and variations of these can be stored in any of 50 non-volatile user memories. The DN780 currently includes ADT, Multi-Tap Echo, Sound on Sound, Straight Delay and Infinite Room; additional programs on EPROMS will be made available as they are developed.

KLARK-TEKNIK ELECTRONICS

For additional information circle #131

TOA INTRODUCES MULTI-PURPOSE STUDIO MONITORS

The top-of-the-line Model 312-ME is a three-way system of symmetrically-arranged components — dome tweeter with diffuser, mid-range cone speaker, and 11-inch woofer — and handles continuous program at 135 WRMS. The front panel provides level controls for high frequency and one for MF adjustment. Designed for primary reference monitoring, the unit is said to provide superb transient characteristics, and a smooth, extended frequency response (50 Hz to 20 kHz).

The 280-ME three-way monitor offers a 60 Hz to 20 kHz frequency response, low distortion, smooth crossover, and wide dispersion. HF response is provided by two tweeters: one soft dome and one polyester dome "super tweeter," both with diffuser (the woofer is a 7.9-inch polypropylene cone). An HF level control is located on the monitor's front panel, and the unit's continuous power handling is 90 WRMS.

The two-way 265-ME is said to be ideal for either primary or secondary reference applications, and handles continuous program at 75 WRMS. Response is a quoted 60 Hz to 20 kHz. Components comprise a 6.3-inch woofer, and a 1.2-inch soft dome tweeter with diffuser.

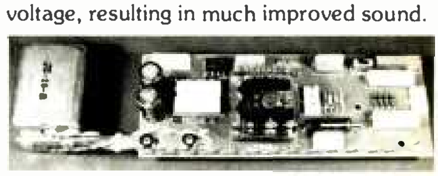
TOA ELECTRONICS, INC.

For additional information circle #132

HARDY MPC-600 MIKE PRE-AMP

The transformer-input pre-amp card directly replaces the stock transformerless mike pre-amp card of MCI JH-600 Series consoles, and is said to provide greatly improved sound over the stock card.

The 990 discrete op-amp is described as being faster, quieter, more powerful and much better sounding than the stock 5534, while the Jensen JE 16-B input transformer provides better matching and proper loading for microphones. Use of the JE 16-B also makes it possible to eliminate input capacitors that are required in transformerless designs to block the phantom-power supply voltage, resulting in much improved sound.



DC servo control of DC offset is also features, and input bias current compensation circuitry eliminates all coupling and gain-pot capacitors for further significant improvement in sound quality.

Pro-user price of the MPC-600 is $184.

THE JOHN HARDY CO.

For additional information circle #133

CROWN MICRO-TECH 1000 STEREO POWER AMPLIFIER

Intended for professional sound reinforcement and studio monitoring, the Micro-Tech has a quoted spec of 1,000 watts continuous average power in mono mode at less than 1% THD, into one or four ohms. A "parallel mono" switch combines the outputs of both channels to make a mono amp capable of 1,000 watts into one ohm. By adding an internal jumper for the "bridge mono" configuration, the user can obtain 1,000 watts into four ohms. In stereo mode, the Micro-Tech provides 250 watts per channel into eight ohms, or 350 watts per channel into four ohms.

Patented Crown circuitry allows extreme voltage swings without putting output transistors in series; this technique is said to provide lower distortion and greater reliability. (Reliability is further enhanced by a redundant power supply.)

The unit uses an "Output Device Emulator Protection" (ODEP) circuit which simulates the output transistors. With this circuit, the amplifier can detect and compensate for overheating and overload. The amp is also protected against output shorts, open circuits, mismatched loads, overall overheating, and high-frequency overloads. Heat sinking and self-contained forced-air cooling system prevent overheating and prolong component life; the direction of airflow can be reversed, if necessary, to work with the rack cooling system.

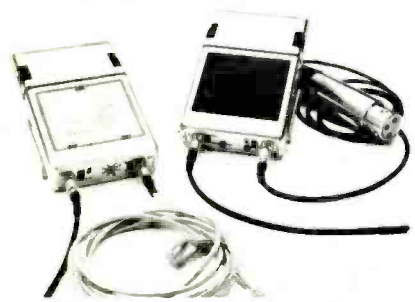
Hum and noise are a quoted 105 dB below rated output (A-weighted), harmonic distortion less than 0.05% from 20 Hz to 1 kHz, and increasing linearly to 0.1% at 20 kHz, delivering 250 watts into 8 ohms, per channel; IM distortion is less than 0.05%, and slewing rate greater than 13 volts per microsecond.

CROWN INTERNATIONAL

For additional information circle #134

NEW LINE OF SENNHEISER WIRELESS MIKES

The SKM 4031 handheld mike transmitter, SK 2012 body pac receiver, and EM 1036 multichannel rack-mount receiver are available in a variety of bandwidths, including UHF high and low.



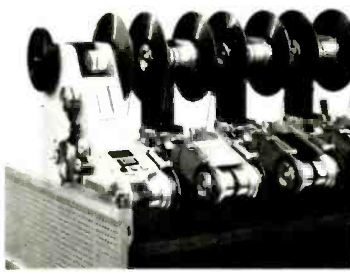
All battery-operated equipment incorporates a DC-DC voltage converter that maintains the operating voltage of the unit as the supply voltage drops, thereby avoiding audio distortion and carrier frequency drift. Powered by three AAA batteries, the units are equipped with special compander circuitry, crystal-controlled oscillators, and select features which vary from model to model.

SENNHEISER ELECTRONIC CORPORATION

For additional information circle #135

MAGNEFAX MODEL 7574 HIGH-SPEED DUPLICATOR

Featuring the common mandrel design which has been in use on all Magnefax duplicators since 1959, the new Model 7574 also incorporates what are described as some of the most needed features for high-quality tape duplication: Digital peak meters with memory hold to provide pertinent level information throughout the entire bandwidth even during the shortest musical passages. The total low-level audio path from the playback heads has been shortened to the minimum, while the use of selected components and plug-in amplifiers is said to ensure total immunity from noise and high-end dropouts.



A digital counter on the control module provides the user with the number of tapes recorded per slave, and the machine will stop automatically when the preset number of passes has been reached.

The loop bin will accommodate up to 1,800 feet of ¼-inch master recorded at 7.5 ips. The conductive nature of the material used for the unit reduces static electricity; the open design also eliminates problems associated with air compression, which often result in poor tape handling.

The three-point plates used for the record heads are said to provide perfect support and stability, and minimize the interactions between the adjustments; heads are long-life/high-output Permalloy for best headroom and low noise. Dual synchronized bias oscillators are used for optimum crosstalk rejection and frequency stability.

The 7574 will produce in excess of 6900 C-45s per 24-hour day using a 16:1 duplicating ratio for optimum quality and low maintenance. Frequency response is a quoted 30 Hz to 15 kHz, ±2 dB, crosstalk from A to B more than 55 dB, and signal to noise ratio within 2 dB of bulk erased tape.

MAGNEFAX INTERNATIONAL, INC.

For additional information circle #136

NEW MC SERIES MONITOR CONSOLES FROM YAMAHA

The new MC1608M and MC2408M monitor mixing consoles feature 16 and 24 inputs, respectively, with all primary inputs and outputs on balanced lines via XLR connectors. Both consoles incorporate the same basic features, with eight master outputs, two auxiliary sends, two fully assignable auxiliary returns, and VU metering of bussess 1 thru 8, plus the two auxiliaries.

The consoles are modular in construction, with blocks of four input channels for easy service when necessary. Each input channel features a pad switch and gain control with peak LED, phase reversal switch; three-band

New Products

EQ with sweep midrange; two post-EQ and pre-fader auxiliary sends; eight rotary master send controls; channel on/off and cue switches with input channel cue priority. All knobs are color-coded between input and output sections to aid identification in low-light situations.

One of the MC Series' unique features is said to be its Input Channel Cue Priority. When an input channel cue switch is pressed, the previous master cue is cancelled. During normal operation, the engineer normally will monitor one of the eight master outputs. If, for example, a feedback problem arises, one or more input channel cue switches can be selected, and the console automatically replaces what was previously being monitored with the new channel's program.

Suggested retail price of the MC1608M is $2,895, and $3,995 for the MC2408M.

YAMAHA INTERNATIONAL CORP.

For additional information circle #138

ELECTRO-VOICE DL SERIES WOOFER WITH EXTENDED LF RESPONSE

The new DL15W and DL18W very-low-frequency drivers are said to provide low-frequency response extension and greater peak output in the bass region. In the same size enclosure, the DLW model provides more extended bass response than its DLX counterpart; in a larger enclosure, the DLW will provide even more extended bass response, EV claims.

The DL15W is a 15-inch LF driver for compact, extended-bass woofer and subwoofer applications, particularly for frequencies above 40 Hz. Specifications include a one-watt/one-meter sensitivity of 97 dB (100 to 800 Hz) and a long-term average power capacity of 400 watts per EIA standard RS-426A. The 18-inch DL18W driver is also for subwoofer application, especially for frequencies below 40 Hz. Sensitivity is 95 dB and long-term average power capacity 400 watts.

"A host of exclusive features contribute to the DL's outstanding performance," explains Jim Long, director of marketing. "EV's unique Thermo Inductive Ring and PROTEF work together to maximize power capacity. The TIR is an aluminum ring that sits on top of the pole piece and provides a major heat-transfer path from the top of the voice coil. PROTEF is a Teflon-based coating applied to the inside diameter of the top plate next to the voice coil; when high power expands the coil, PROTEF lubricates the contact and inserts electrical insulation between the coil and the top plate."

ELECTRO-VOICE, INC.

For additional information circle #139

dbx MODEL 166 COMBINED DYNAMICS PROCESSING UNIT

The new Model 166 is three dynamics processors — noise gate, compressor/limiter and peak clipper — incorporated in a single package; all three functions can be used simultaneously. The noise gate threshold is variable over a wide range, while its attack time is brief to allow the complete transient at the beginning of the signal to come through. Release time is controlled by two proprietary dbx RMS detectors (one fast, one slow) selected on a front-panel switch.

The unit's variable compressor function incorporates dbx's "Over Easy" compression, which is described as being gentler and smoother than the typical "hard-knee" operation of most compressors. Compression ratio varies from 1:1 (off) to infinity.

Peak stop, the company's "Intelligent Clipper" designed originally for the Model 165 A compressor, sets an absolute limit on peak-output level, so that the signal cannot cause overmodulation, or damage a PA system. The amount of gain reduction and compression is displayed on front-panel LEDs.

Rear-panel connections are supplied for connecting an equalizer or other signal processor into the unit's side chain to provide, for example, frequency-dependent compression.

Retail price of the Model 166 Dynamics Processor will be under $600, with March availability.

dbx, INC.

For additional information circle #140

TANDBERG INTRODUCES FIRST PROFESSIONAL REEL-TO-REEL DECKS

The new TD50 Series of $\frac{1}{4}$-inch transports feature a transport design in which all mechanical components are mounted directly on a 10mm-thick place of Alcoa Alca Plus aluminium alloy. This alloy is molded to extreme tolerance through a proprietary process, maintaining a perfect flatness in its "memory," and thus has the ability to return to its original shape if ever it is deformed by stress or temperature, Tandberg says.

A major benefit of such a transport design is that it also eliminates the need for a chassis, which can inhibit machine repair or adjustment; all transport mechanics and power electronics are mounted directly on the rear of the Alca slab. Releasing two screws allows the transport to swing back for immediate access or removal of any part without having to disturb other assemblies.

A built-in microprocessor controls the five-digit, real-time (elapsed) counter and functions such as cue/rewind, variable wind speed, tape dumping, return-to-zero, return-to-start, set cue/search cue, fader start, etc., and also enables optional SMPTE timecode and frame sync capabilities.

Other features include: built-in tape cutter for editing; fast, easy access to the head block module with the removal of a single screw; patented tape guide system utilizing sold ruby cylinder tape guides for minimum-wear performance; a three-motor transport, including a direct-drive capstan motor; three tape speeds (15, $7\frac{1}{2}$ and $3\frac{3}{4}$ ips); and built-in monitoring speaker and amp.

TANDBERG OF AMERICA, INC.

For additional information circle #141

TIMELINE ANNOUNCES LYNX TIMECODE MODULE

The new LYNX timecode module incorporates the three elements of a complete timecode system in a single half-rack enclosure. Each module contains an independent timecode generator, a wide-band timecode reader (to 60x speed), and a transport synchronizer with built in parallel interface. To function as a high performance chase synchronizing system, one module is connected to each controlled transport. Up to 32 modules/transports can be on line simultaneously.

The system is said to provide the unusual ability to freely select any machine as the current timecode "master," and to take machines on- and off-line without regard to hierarchy. Also featured is an RS422 serial port for external computer control, and for interconnect to the forthcoming LYNX timecode controller.

Pro Net price of the LYNX unit is $2,450.

TIMELINE, INC.

For additional information circle #142

ENHANCED SYNCLAVIER KEYBOARD FROM NEW ENGLAND DIGITAL

Standard with every Synclavier Digital Music System, the Synclavier now offers a 76-note, velocity and independent aftertouch, user-programmable keyboard. The system also features a 32-track digital memory recorder, pitch and modulation wheels, optional breath controller, and provisions for SMPTE and MIDI.

Along with the above features, the Synclavier's keyboard is said to feature a greatly expanded real-time effects section, which permits the user to create 192 different types of patches.

NEW ENGLAND DIGITAL

For additional information circle #143

UHER 160 PORTABLE STEREO CASSETTE

The Model 160 can be powered by six dry cells, nickel-cadmium rechargeable batteries, 12-volt car batteries, and 110/220/240, 50/60

Hz current. The recorder also features both Dolby B and C noise reduction; three built-in speakers for on-site monitoring; separate right and left level controls and switchable AGL with two time constants; and twin peak-reading meters.

Transport features include a front-loading cassette compartment; three-way tape selector; line and mike inputs, as well as line, and monitor outputs, and long-life Sendust heads.

Optional accessories include facilities for film dubbing, sync sound (Uher 160AV) and remote control.

Suggested retail price of the Uher 160 is $905.

UHER OF AMERICA

For additional information circle #153

UPDATES FOR SOUND WORKSHOP SERIES 34 CONSOLES

The following features are now supplied as an integral part of the Series 34:

- Monitor solo — a locking pushbutton with indicator LED added to the stereo mix section sections allows soloing of the post monitor pot signal.
- Fader-flip — A locking pushbutton with indicator LED in the fader area permits the exchanging (reversing) of the monitor signal with the channel signal, to allow VCA bypass, and monitoring of the multitrack on the main faders. (Reversal is done on a channel-by-channel basis to provide maximum console flexibility.)
- Performance enhancements — further optimization of critical circuits is said to have increased the dynamic range and signal-to-noise ratio, and decreased crosstalk by up to 5 dB; stereo panning networks have been revised to yield sharper cut-off characteristics.
- Mechanical meter option — may now be specified in place of, or in addition to, the current hi-resolution LED metering.
- Serial console interface option — the new SCI allows the console to be controlled by a video editor or other control computer.
- Auxiliary monitor mixer — available in 8-, 16-, and 24-channel configurations for mounting in the meter bridge, each channel features level, panning, solo, and mute functions.

SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS

For additional information circle #154

APHEX ANNOUNCES MODULAR AURAL EXCITER

The new Model 900B is designed to fit Aphex rack systems, or dbx F900 rack systems.

"By providing modules which will fit in existing rack systems, we are giving recording, post production, broadcast and sound engineers the opportunity to add the psychoacoustic enhancement of the Aural Exciter to every audio channel," exlained Aphex president, Marvin Caesar. "Until now, engineers had to compromise by using an Aural Exciter on the final mix or they had to use banks of two-channel Aural Exciters."

Suggested professional user price of the Model 900B is $295.

APHEX SYSTEMS, INC.

For additional information circle #155

MIDI UPDATE

— continued from page 44 . . .

store and load sounds to and from the Apple's floppy-disk. A total of close to 850 sounds can be stored on one disk and the envelope parameters can be printed both graphically and numerically, as well as all the DX-7's parameters. (For many producer's using our studio, paying $8.00 for a diskette makes much more sense than a hundred bucks for a DX-7 RAM cartridge.)

The new **Roland Super Jupiter**, an eight-voice, analog velocity synth, is among the few first rack-mounted MIDI synthesizers, and has no keyboard of its own. (This rack-mounted style of synthesizer provides the player/owner with a choice of preferred keyboard type.) The Super Jupiter's light weight and size standard 19-inch by two rack spaces make it ideal for the studio. But that's not reason enough to buy one — its unique sounds, with great factory presets, and capability with all MIDI keyboards are definitely the deciding factors. The device stores 64 sounds internally and, with the external cartridge inserted, you can choose from two more banks of 64 (192 sounds total). Sounds are edited *a la* DX-7 style — one slider that can access all the parameters — and the back-lit LCD display shows where each parameter was numerically before, compared to where you have changed it, in reference to the factory preset. As an option, you can purchase a separate programmer with all the knobs found on most analog synthesizers.

Roland also plans a software update for the **MPU-401**, a computer interface box that allows a host computer to function as an eight-track MIDI sequencer. The MPU-401 normally comes complete with software for either the Apple II, II+, IIe or the IBM PC, XT (and possible the new AT, for it has not yet been tested). The new MPS software, which is only available for the IBM PC family, is a music printing/scoring program with "on screen" editing of any bar of music. When editing, you call up any one bar from any of the eight tracks at a time. The entire screen shows that one bar (Piano clef) and you can then type lyrics or, by moving the cursor to a particular note, change its value or pitch notation on the staff. MPS scores can be printed via a Epson FX-80 or compatible printer.

The main screen/menu shows all eight tracks at once; each track is then displayed as little empty boxes from left to right, each empty box equalling one bar. As data are written onto a track the boxes become filled, which shows where there is music written, and where there are rests in the composition. You can also cut and paste any bar(s) to any other track or to the same track, repeat it for any length, or delete it. The MIDI channel assignment for each track can be assigned later.

. . . continued overleaf —

Roland Super Jupiter synthesizer controller

Roland SAX-80 Sync Box

MIDI UPDATE
. . . continued —

In the sampling department, **E-Mu Systems Emulator II** provides up to 17.6 seconds of sampling time at 16.5 kHz bandwidth. The keyboard can be divided as many times as you please, and you can even sample on every note. The Emulator II is totally MIDI-compatible, and its five-octave keyboard is velocity sensitive. The amount of control one has over a sampled waveform is quite staggering, especially since the unit has four sliders that can be assigned to a parameter found on any analog synthesizer — i.e., VCA ADSR, filter, filter ADSR, and so on — which speeds up the editing of sampled sounds. There is even automatic looping software to make glitchless loops, without the need of a monitor. The E-II has eight individual outputs, and each of the multisamples can be routed to any of the eight outputs. Up to 99 presets of filter, VCA, LFO, transpositions, and different outputs are available, and the keyboard comes with a complete on-board sequencer/ arpeggiator. The sequencer will record velocity, pitch-bend, modulation and sustain pedal status, and is totally compatible to record and play back all other MIDI keyboards.

The real deal again is that the Emulator II sounds so real that it has to be heard and played to be believed.

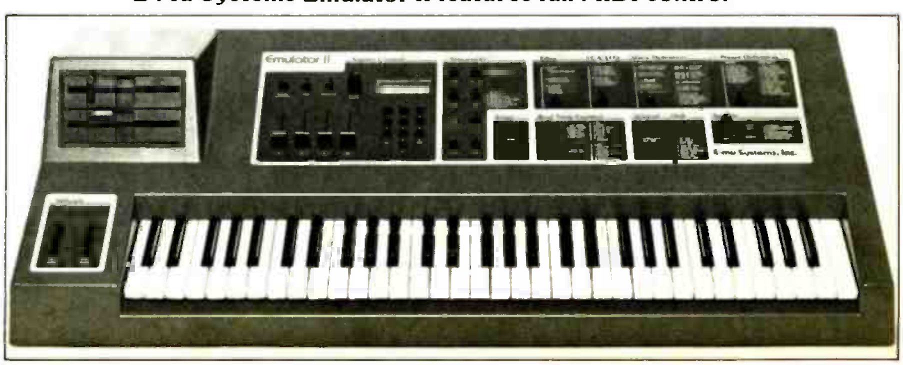
Ensoniqs has introduced the **Mirage**, a low-priced alternative to the Emulator II (at about a third the price), and is somewhere in between the original Emulator and the Emulator II. It has a five-octave velocity-sensitive keyboard that can be split with up to three sounds per keyboard half. Sampling bandwidth is as follows: 15 kHz for two seconds, 8 kHz for four seconds, and 4 kHz for eight seconds (which is the maximum sample length). The keyboard contains an on-board sequencer that records pitch bend, modulation, velocity, and the sustain pedal. It also has multi-overdub ability, and can record other MIDI equipment. The Mirage uses a Sony 3½-inch disk drive for storage; software will be made available in the future to interface the unit to the Apple Macintosh for on-screen editing purposes.

Europa Technology has updated the MIDI capability on the **PPG** and **PPG/Waveterm** synthesizer systems to allow the PPG to interface with all other MIDI equipment. Europa has also introduced the **EVU**, a rack-mounted PPG that features all the same functions plus MIDI In/Outputs. Also introduced was the **PRK** keyboard that can be used as a master MIDI keyboard, or can be filled with cards given the same capabilities as the PPG. All three units — the PPG with or without Waveterm, the EVU, and the PRK — can be used together to provide a total of 24 tracks on the digital sequencer, and more voices with different sound patches.

Fairlight has added a MIDI In/Out card to replace the analog interface card on the **CMI**. With the new MIDI card installed you can trigger a CMI and/or program Page R from any MIDI equipment; in return Page R can also play back other keyboards.

Linn Electronics has just released the **Linndrum 9000**, which has 18 velocity-sensitive drum pads, and is described as the first drum machine to record velocity information. In addition, the 9000 can be triggered by external pads — for example, Simmons — to record a live drummer's performance. It also contains a digital MIDI sequencer that includes 99 sequences, with 32 tracks per sequence. The 9000 is reportedly designed around an IBM PC computer, and there will be many extra options offered, including a Sony 3½-inch disk drive, an additional six trigger inputs, and a sampling option. the Linndrum 9000 can best be described as being like no other drum machine sequencer you've ever encountered before. □□□

E-Mu Systems Emulator II features full MIDI control



— THE COVER —
— continued from page 5 . . .

and mono Ampex recorders, and a custom-built console made for us by Jeep Harned of Music Center, Inc. in Fort Lauderdale, Florida. This was Jeep's first complete console construction job; in 1964, all consoles were custom made to order. We had the cleanest, best-sounding studio in town. We followed The Tams production with another best-seller by Billy Jo Royal: 'Down in The Boondocks.' We were off and running!

"By 1968 we expanded to eight-track recording — we were the first studio in Atlanta with eight tracks. In 1971, we moved to a new location with a larger studio designed by Tom Hidley and George Augspurger, and installed the first 16-track recorder in Atlanta. We also equipped this new room with another custom board constructed by Jeep Harned (by then MCI was well on its way) and really jumped ahead of the other studios by adding an equalized speaker system in the control room, using the Boner System and tuned by Tom Hidley. This studio today is unchanged, except for the addition of a drum booth in 1974, and it's still one of the best-sounding rooms around. We plan to update the control room acoustics, but the studio will remain untouched.

"During this period we recorded hits by Lobo, Jim Stafford, Goose Creek Symphony and the soundtrack for the movie *Deliverance*, and the single hit 'Dueling Banjos.' In 1975, we installed 24-track, again the first in Atlanta, and in 1976, we added a new MCI 500 console with automation: a first again — no one in New York or Nashville had automation. We had automation serial #002 from MCI; the first went to CBS in London. From 1976 to the present we've made many equipment upgrades: new monitor plus adding an additional 24-track machine and SMPTE generators and synchronizers.

"Today the majority of our work is dual-24 at 30 ips with no noise reduction. We started experimenting with dual-24 nine years ago, when Isaac Hayes needed more tracks; it was crude then, but it worked. Because of this experience, we were ahead of the trend in locking video and audio machines.

"With this history of Master Sound, I've tried to point out that we have always been at the forefront of technology, and have been first with new products and concepts when they've proven to be better.

"Every major upgrade was a gamble at the time we did it. There have been some nervous times when we have been ahead of the trends, and have had to pioneer a new concept before it was standardized, but the Atlanta market has been good to us and we have grown with it.

"Atlanta is a large regional, diverse and expanding market. It's considered marginal when compared to New York and L.A. in recording for the Record Industry. Since we've recorded some 30 Gold records, it's obvious we've had some opportunities in this arena, and we've been successful when the ingredients were right. We were prepared with the right equipment and expertise when we hthe opportunity. Atlanta is an ad agency center, with every major national agency represented by a branch office, and some very large independent agencies with national accounts; this has given us a base of commercial music recording, spot production"and high-speed duplicating. It is demanding, professional work that is very competitive. We found early in business that commercial recording was steady, and balanced the fluctuations in record production. Comm"rcial music clients also appreciate and demand state-of-the-art equipment. This also affects our decision making in equipment purchases.

"So, about about a year and a half ago, we took stock of our current equipment inventory and tried to project where we were going and to forecast trends that would affect our future business. We decided that level-set automation alone was not enough for the future business. Total automation was promised in an all-digital console, but the technology and the components to make this possible or affordable in the near future were not available. In April, we decided on SSL. We would have level-set automation, plus Total Recall and computer aid in performing all the tasks required in multimachine lockup.

"A purchase of this size is *not* for the timid. It's not easily affordable for a studio like ours, which is strictly a rental-service facility. We have no in-house production that generates royalties to help pay for equipment. We projected a rate increase would be necessary and, with increased capabilities, felt we could justify it. It would be another five-year plan, just like previous upgrades. Those improvements in the past worked for us, and we had enough faith in ourselves and the market to make the purchase. That's our thinking on the acquisition.

"Financing? We handled the financing through our bank. Remember, we started thinking about this over a year ago. At that time we discussed buying a new console with the bank. We didn't know then what we were going to buy or exactly when, but we've always discussed our plans with the bank right from the start. We keep them appraised of what we're doing . . . who we are doing business with and what our goals are for the future. It's time well spent. Then when we decide to ask for a loan to make a major purchase, it's no surprise to them.

"The SSL is in . . . we think it's great . . . and our clients think it's great." ■■■

News

APPLIED RESEARCH AND TECHNOLOGY PURCHASES MXR BRAND NAME

The new corporation is comprised of engineers and managers that originally were all employed by MXR Innovations, Inc. The principals of ART are Richard Neatrour and Tony Gambacurta, both of whom are said to have extensive backgrounds in both digital and analog signal processing; John Langlois, a specialist in manufacturing operations and logistics; Phil Betette, President of ART; Pete Beverage, national sales manager; and Terry Sherwood, controller.

The members of ART are described as being strongly committed to the support and futherance of the Audio and Digital Sound industries through quality products using the latest available technology, and welcomes dialog with all interested end-users, studios, and sound companies.

ART purchased the rights to the MXR brand name and will carry the MXR identity through 1984.

UHER RE-OPENS U.S. SALES AND MARKETING OFFICE

The new sales, service and marketing office, located in Los Angeles, will be headed by John A. Belgiorno, president, and George A. Rose, VP marketing.

"Uher has re-dedicated itself to the American market," says Rose. "By centralizing sales, marketing, parts and service under one roof, we hope to provide a vastly improved and centralized source for Uher's entire domestic effort." Uher of America will handle both professional and consumer product lines.

The company's address is: Uher of America, 7067 Vineland Ave., North Hollywood, CA 91605. (818) 764-1120.

SONY PROVIDES GRANT TO SPARS FOR EDUCATIONAL TESTING PROGRAM

The $55,000 grant to the Society for Professional Audio Recording Studios will be used to establish an educational testing program for aspiring audio technicians. The program, being developed by the Educational Testing Service of Princeton, New Jersey, is intended to help better define job performance standards for audio technicians, and to assist in developing educational curricula.

AEG-TELEFUNKEN APPOINTS QUAD EIGHT/WESTREX EXCLUSIVE U.S. DISTRIBUTOR FOR TAPE MACHINE LINE

According to Rudiger Barth, director of sales and marketing for AEG-Telefunken, the appointment of Quad Eight/Westrex represents a change in marketing strategy for the company. "We looked for an organization that was truly dedicated to the professional audio industry," says Barth. "In Quad Eight/Westrex, with its tremendous investment in both analog and digital research and development, as evidenced by their recent product introductions, I feel that we made an excellent decision."

Quad Eight/Westrex now has manufacturing facilities in both Los Angeles and London. "We recently opened a sales office in Nashville, where Dave Purple serves as director of sales, Eastern United States," reports Cam Davis, president. "I feel that our new association with AEG-Telefunken will be of

. . . continued on page 177 —

Classified

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11. I certify that the statements made by me above are correct and complete. Martin Gallay, President/Publisher

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EVERYTHING MUST GO!!

MCI JH-536 console w/everything. The following MCI tape machines: JH-24 w/ AutoLocator III + remote, JH-114 w/ Autolocator II + remote, JH-45 AutoLock, JH-110-4 w/2 track heads, JH-110A-2 + remote. ALSO, 4 Westlake studio monitors, many BGW 500 & 250 power amps. Microphones, Outboard Gear & Test Equipment. 9 ft. Kawai Concert Grand Piano. This equipment has been handled with kid gloves. For a more complete listing and prices call (305) 361-3367.

FOR SALE

API console 24/8, EMT 140, Neumann U 47, Dolby 16 rack.
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Also: Revox PR99's, Fostex *Pro*, AKG, ADA Efx., Symetrix, Klark-Teknik, Micmix, NEI, AB Systems.
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News

— *continued from page 173 . . .*
mutual benefit to both companies."

The company also has relocated its intetional headquarters to: 225 Parkside Drive, San Fernando, CA 91340. (818) 898-2341; the Telex number will remain 662446.

TELEX PURCHASES ALTEC CORPORATION

Telex Communications, Inc., a wholly owned subsidiary of The Telex Corporation, has reached an agreement in principle with Altec Corporation, whereby it will acquire substantially all of the assets and business of Altec for a purchase price exceeding $12 million. The purchase price is to be paid principally with cash and the assumption of certain specificed liabilities.

Consummation of the transaction is subject to the execution of a definitive agreement between the parties, the approval of their respective boards of directors, and the confirmation of a plan of reorganization by the U.S. District Court supervising the reorganization of Altec under chapter 11 of the bankruptcy act.

Ansel Kleiman, chairman and CEO of Telex Communications, stated: "This acquisition will complement and broaden our involvement in the field of audio communications to which we are dedicated. We look forward to pursuing this market aggressively in the future."

AES CONVENTION/EXHIBITION SCHEDULE; NOW ONE U.S. SHOW PER YEAR

The Governors of the AES have formulated the following Convention/Exhibition policy for the U.S.:

- There will be only one Convention/Exhibition per year;
- The event will be held in the Fall of each year;
- The event will alternate between the East and West Coasts; and
- The event will alternate in location with the SMPTE Convention.

From 1986 the schedule will be: 1986 Fall — West Coast; 1987 Fall — East Coast; 1988 Fall — West Coast; 1989 Fall — East Coast; 1990 Fall — West Coast.

The AES will cancel exhibition space booking for the Spring of these years, and no other exhibition opportunities will be offered in North America by the AES. All exhibitors will be polled annually covering their views on future exhibition policy.

Regarding 1985, the governors report that, taking account of space availability, they considered alternatives of having no West Coast Convention/Exhibition for three years or no East Coast Convention/Exhibition for three years. Neither alternative was considered to be in the long-term interest of either exhibitors or members. It has been decided, therefore, that for 1985 *only* there would be two Convention/Exhibitions: Spring, Anaheim; and Fall, New York.

- The dates of the AES 78th Convention, to be held at the Disneyland Hotel, Anaheim, California, are Friday, May 3, thru Monday May 6. Milton "Bill" Putnam will serve as chairman emeritus, Dean Austin as convention chairman, Bart Locanthi as papers chairman, and *R-e/p*'s Laurel Cash as workshops programs chairperson.

A/T SCHARFF RENTALS INTRODUCES E-MAIL ORDERING SYSTEM

Premiered at the recent AES Convention in New York, A/T Scharff's new service uses the IMC Systems Network to allow clients to order the company's equipment from virtually anywhere around the world via a personal computer.

The system, developed by company president Peter B. Scharff and IMC's Chris Coffin, is the first such system on the IMC Network. Both Scharff and Coffin expect it to be the first of many similar uses of the network.

With the new electronic ordering system, a record producer in London can book equipment for an upcoming session in New York, for example, or a touring band can quickly replace a piece of equipment while out on the road. Not only is the system described as easy and efficient, but IMC's international computer network is said to be less expensive than long-distance or overseas phone calls.

Boasting more than 1,300 users across the U.S., Europe, Australia and Japan, IMC's clientele includes most major rock and roll bands, record companies, producers, film and video companies, production and tour managers, sound and lighting companies, as well as the SPARS membership.

- *R-e/p* can now be contacted via IMC by EMail to REP-US, IMC 822 — *Editor.*

MICROPHONES STOLEN FROM L.A. CITY COLLEGE

The following five microphones recently were stolen from the Los Angeles City College.

- Neumann U-87, s/n 37654 (also,

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News

School Asset #176537).
- Neumann U-87, s/n 37625 (School Asset #176538).
- Sennheiser MD441-U, s/n 019823 (School Asset #176848).
- Electro-Voice RE-20, no s/n (School Asset #83261).
- Electro-Voice RE-20, no s/n (School Asset #83263).

All microphones are engraved "L.A.C.C. Radio-TV-Film Dept."

If any *R-e/p* reader comes across these stolen models, please contact Norman E. Cobb, chief engineer, Radio-TV-Film Department, Los Angeles City College, 855 North Vermont Ave., Los Angeles, CA 90029. (213) 669-4000.

— News Notes —

According to West Coast accounts manager Joni Lyman, **U.S. Concord**, a company specializing in the leasing of professional audio and video equipment, has moved to a larger facility located at: 303 N. Glenoaks Blvd., Suite #670, Burbank, CA 91502. (818) 841-5050 . . . Further emphasizing its committment to sales and service of high technology musical instruments, recording and broadcast systems, and computer interfaces, **Leo's Professional Audio, Leo's Music** and **Leo's Drums** have merged to form a single integrated company. The new company will be called **Leo's Audio and Music Technologies**.

— People on the Move —

- **A. Franz Witte III** has joined the **Ampex Magnetic Tape Division** as manager of market research and planning, where he will create and implement a series of Division market analysis programs in all areas and, based on the results of these programs, will develop strategic models for future Division growth. Also, **Wm. Bruce Pharr** has been appointed manager of marketing communications for the Division, with responsibility for developing and implementing a comprehensive marketing communications program for the division's professional audio, video and instrumentation tape lines.
- **Lance Korthals** has been appointed director of marketing and sales for **Lexicon, Inc.** Prior to joining the company, Korthals had been director of marketing and sales for dbx, Inc., and has held positions with Altair Corporation, Interactive Systems, Inc., and Head Sound, Inc.
- **Dan Abelson** has been appointed VP of sales and marketing at **Turbosound, Inc.** Since 1982, Abelson served with the Audio Marketing Group, Limited and was responsible for representing Turbosound products during their introduction into the U.S. pro market. He will be responsible for coordinating U.S. sales activity, in addition to broadening the company's expanding market.
- **Bill Raventos** has been named microphone product director for **Crown International**, where he will be responsible for microphone product definition, design input, field evaluations, and working with reps and end users to determine microphone-related needs. He is also planning a series of seminars for dealers and end users on the company's microphone line. For three years Raventos was director of technical services for Ringling Brothers Barnum & Bailey Circus at their Circus World theme park in central Florida. For 10 years, he was known for his work at Electro-Voice in marketing and new product development of professional microphones and monitor speakers. Later he was responsible for development and marketing of acoustic test equipment at Ivie Electronics for four years.
- **Paul Yurt** has joined **Soundcraft Electronics'** New York office. He was formerly with Neve International, London, England, and most recently was employed by Digital Effects Corp., a New York-based computer animation facility. Yurts' responsibilities will include overseeing the technical sales of the company's new TS-24 console. Also joining the New York operation is **Phil Wagner**, formally an engineer at Soundworks studio. His responsibilties will include local technical sales support on the dealer and consumer levels. In addition, **Patty LaMagna**, formally of Frontier Booking International of New York, has joined the NYC staff. In addition, **Steve Smulian** has been named senior technical engineer. Formerly of AVE Systems, Smulian will focus specifically on the Series 4, Series 2400 and TS project.
- **Ray Kirchhoefer** has joined the **Electro-Voice** engineering team as engineering project manager/microphones, where he will be responsible for defining and developing new microphone products and product line concepts. Prior to the appointment, Kirchhoefer was a project engineer at Shure Brothers, Inc. ■■■

OTARI MTR-12 REVIEW — Late Reply from Peter Butt

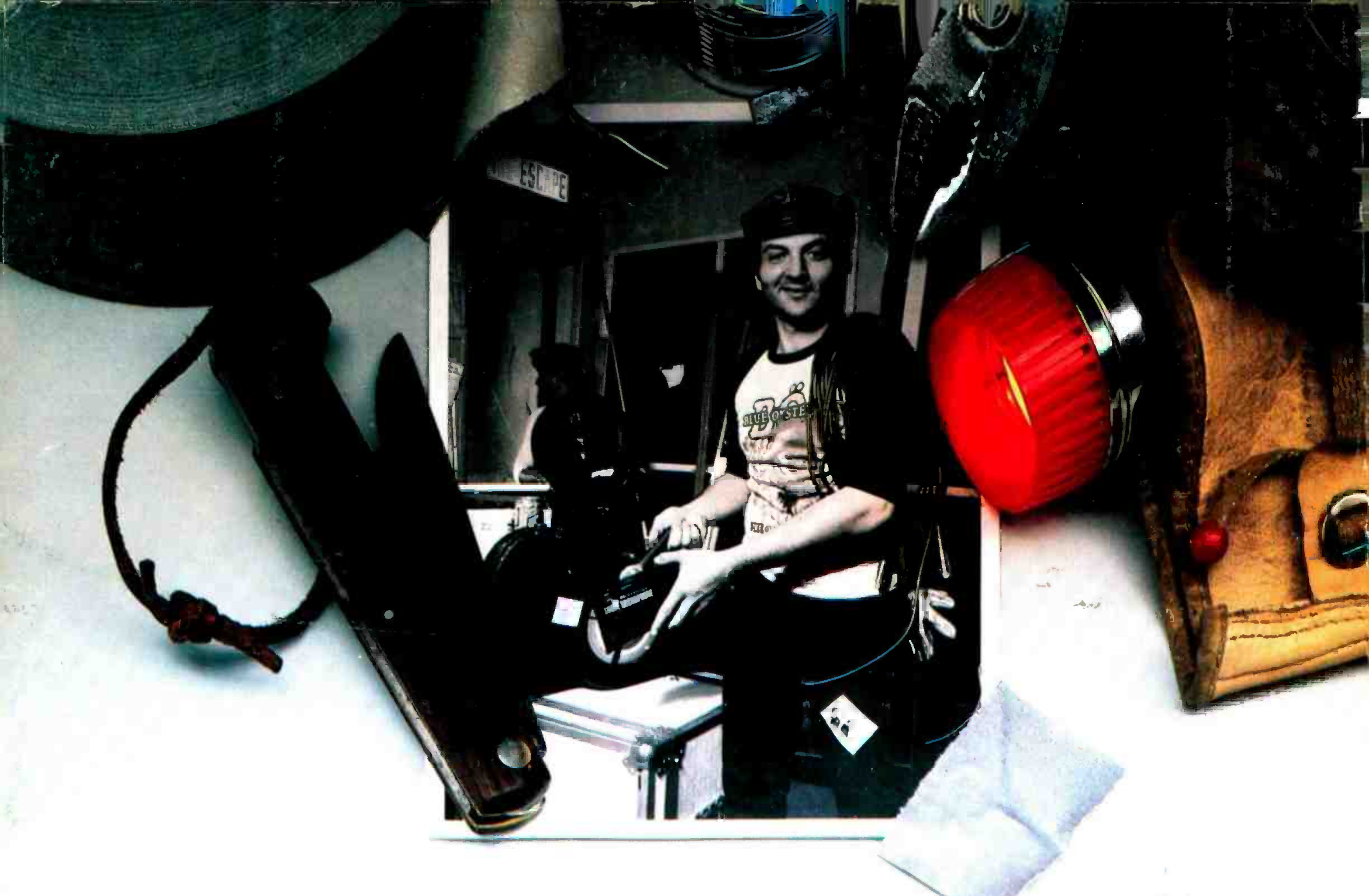
I appreciate Mr. Carlstrom's comments. ["Letters," page 6.] The details of the MTR-12/10 bearing dimensions and their corresponding flutter frequencies are valuable service information that can be profitably applied to obtaining peak performance of the MTRs and all other tape transports.

Mr. Carlstrom's speculation disregards the fact that I mentioned that the offending Otari MTR-12 flutter component at 30 ips is observable only in the replay of the flutter test signal; it is not observable during reproduction of the flutter test signal during recording. Flutter observed during direct replay of a flutter signal is commonly used by myself and others to serve as an indication of relative flutter performance while the deck of interest is subjected to minor fine adjustments. During the testing of the MTR-12, I attempted to improve the machine's flutter performance by changing the pinch-roller tension, and other adjustments, with no notable change.

I believe that I was correct in inferring that the wavelength of the 27 Hz flutter component corresponds to an integer multiple of the record/play gap separation, because it could not be observed in record/play conditions. I am indebted to Dale Manquen for this observation.

The instruction manual supplied with the MTR-12 gives no counsel regarding the fine tuning of the deck for optimal performance. As mentioned in the equipment assessment, it would profit the user if such guidance were supplied with greater candor. □□□


THE "THIRD" TRACK
FEATURED ACTOR: A810 TC
TAPE FORMAT: 1/4"
AUDIO TRACKS: 2 (STEREO OR 2 TRACK)
CODE TRACK: THE "THIRD" TRACK
CODE FORMAT: SMPTE/EBU
CROSSTALK CODE-AUDIO: >90dB
OFFSET: ZERO
OFFSET COMPENSATION: µP-CONTROL DELAY LINE
COINCIDENCE: EXTREMELY PRECISE



The world's least conservative profession has maintained one rigid tradition. The SM58.

In an industry that discards electronic products like ice cream wrappers, the SM58 and its close cousin, the SM57, have remained the overwhelming choice of rock, pop, R & B, gospel and jazz vocalists for the last 16 years.

Why?

Simply because there is no sound quite like the SM58 sound. Its punch in live vocal situations, coupled with a distinctive upper mid-range presence peak and fixed low-frequency rolloff, give it the trademark quality no other manufacturer can imitate, although others have tried.

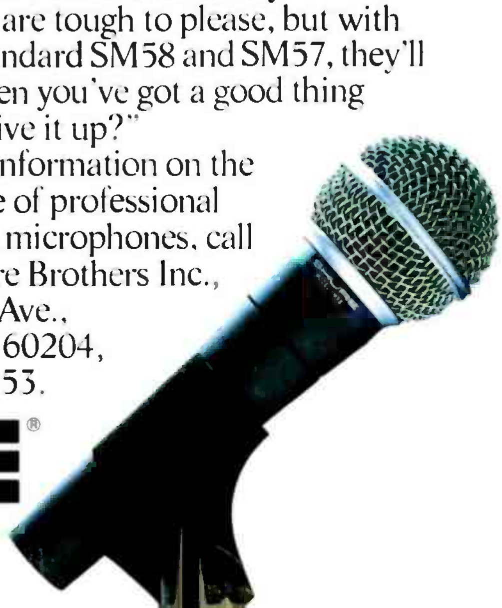
And to protect that sonic perfection, the SM58 is extraordinarily tough. Even six-foot drops on hardwood floors won't faze it. Ask any roadie who has used–and abused–one.

Performers the world over favor the weight and balance of the SM58, especially in hand-held situations. Even the finish is totally professional–a non-glare grey that looks as great on stage as it does on camera.

The crispness of the closely related SM57 enhances musical instruments the way the SM58 handles vocals. Beautifully.

Musicians are tough to please, but with the world-standard SM58 and SM57, they'll tell you, "when you've got a good thing going, why give it up?"

For more information on the complete line of professional performance microphones, call or write Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, (312) 866-2553.



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