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April 1984
Volume 15 — Number 2
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AUDIO PRODUCTION FOR BROADCAST
Pages 144-182

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For additional information circle #2

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In Japan, where today's digital audio studio recorders are developed and manufactured, the World Class Studios are choosing the Mitsubishi X-800 32-Track over the DASH format.

Look to Nippon-Columbia Studios. No other studio organization in the world can equal the digital audio research and development capabilities of Nippon-Columbia. They developed and built one of the first 4-channel digital audio recorders in the world just to mention one of many projects. When it was time to choose a digital multitrack machine, exhaustive tests were performed. Nippon-Columbia selected the Mitsubishi X-800 32-Track. Look to Toshiba-EMI Studios, one of the largest and most famous studio complexes in Tokyo. Their commitment is to produce the finest and best sounding recordings, in the EMI tradition. Which digital multitrack machine did they choose? The Mitsubishi X-800 32-Track, of course.

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The Mitsubishi 32-Channel Standard on one inch tape can reliably be expanded to double density 64-Channel. As a matter of fact, the Mitsubishi error correction system was developed to support a much higher recording density making 64-Channel Mitsubishi double density more reliable and more compatible over long term than the currently proposed DASH double density 48-Channel format. Mitsubishi has the thin film head technology to make 64-Channel double density anytime. When will we do it? Probably if and when the DASH supporters deliver 48-Channel. You see, our future depends upon us protecting the World Class status of our customers. If you need it, we'll go 16 tracks ahead of DASH. The Pyramid has been divided. You have to choose the level in which your studio will operate in the future. If you desire World Class reputation, we invite you to look into the Mitsubishi X-800 Standard. Please call or write.

32
64

"WE NEED 32-TRACK IN OUR STUDIOS. 24-TRACK IS JUST NOT ENOUGH."

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For additional information circle #3

RECORDING ENGINEER/PRODUCER

— the magazine to exclusively serve the **RECORDING STUDIO** and **CONCERT SOUND** industries . . . those whose work involves the **engineering** and **production** of commercially marketable product for:
- **Records** and **Tape**
- **Film**
- **Live Performance**
- **Video** and **Broadcast**

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

"RECORDING Engineer/Producer"
(ISSN 0034-1673)
is published six times a year by GALLAY COMMUNICATIONS, INC., 1850 Whitley Avenue, Hollywood, California 90028, and is sent to qualified recipients in the United States. One year (six issues) subscriptions may be purchased at the following rates:

United States (Surface Mail) **$18.00**
United States (First Class) **$23.00**
Canada **$23.00**
Foreign **$35.00**
(Foreign subscriptions payable in U.S. funds only by bank check or money order.)

Second Class Postage paid at
Los Angeles, California
and Additional Post Offices.
Send address corrections to:
RECORDING Engineer/Producer
P.O. Box 2449
Hollywood, California 90078
(213) 467-1111

April 1984
Volume 15 — Number 2

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Control Room A at **Soundworks Digital Audio/Video Studios,** New York City, with engineer **Roger Nichols** at the facility's Solid State Logic SL6000E console. Nichols, whose article detailing the various stages involved in preparing digital tapes for CD mastering begins on page 40, has extensive experience working with the 3M Digital Mastering System and Sony PCM-1610 processors. Photography by Lionel Freedman.

For additional information circle #4

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views letters news

VU METERING

from: **Oliver Berliner**
Audio International, Inc.
Beverly Hills, California

John Roberts' otherwise excellent piece on the vu meter in the series "Exposing Audio Mythology" in the December 1983 issue is flawed by just a couple of misleading statements.

He claims, rightfully, that most of today's level-monitoring meters don't meet the "20-year-old standard." Whereas the standard that he's referring to — he mentions USA C-16.5 — was apparently promulgated in 1961, he overlooks the fact that this is merely a rehash of a standard for the vu meter that was issues at the time of its invention by Mr. John Miller of Weston Instruments in 1938, which may be before Mr. Roberts was born. Weston, then Amercia's premier meter manufacturer, had been requested to create a standard volume indicator by CBS, NBC and AT&T. This is all described in my 1959 *Audio Engineering* magazine article, "Uses & Abuses Of The VU Meter," which became the standard treatise on the subject.

The vu meter's ballistic characteristics served the industry very well for decades. However the meter's "action" is often deemed inadequate for use with the rock music of today. By the way, please note my use of the lower-case letters for "vu," which is a part of the original standard, although most writers and editors fail to comply with this in their publications. The buff scale color is also specified, as well as the red and black portions. I further deplore the fact that today, contrary to the intention, the so-called "A" scale is used on virtually all equipment — an improper practice initiated by Ampex *on* its early tape recorders, and copied now by virtually everybody — whereas the "A" scale (vu markings above, percent modulation below) was created for use on gainsets and other test equipment, and the "B" scale was designed for program level monitoring, and thus the majority of the applications.

Another lesser known characteristic is that the 3,900-ohm meter impedance is designed to be 12½ times the impedance of a terminated 600-ohm line when a 3k6-ohm-resistor is in series with it; 12½ times the line impedance being the recognized minimum impedance for a bridging device. Further, the effective 300-ohm impedance of a 600-ohm terminated line, when added to the 3k6 resistor equals 3k9, which is exactly the impedance of the meter itself, and which the vu meter wants to "see" for perfect operation.

John Roberts replies:

Thank you for the historical footnote on "vu" meters. While I agree that the "B" scale appears more logical for most applications, it may have fallen from favor due to our tendency to think in decibels, especially when setting record levels.

Rather than rock music, which is not particularily dynamic, I suggest close miking and multitracking have fostered the need for faster ballistics.

I avoided discussing meter termination/impedance for the sake of clarity. Anyone building a system should study the referenced specification. If you just read the meters don't worry about it. □□□

AMBISONIC RECORDING

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

With reference to the article "Ambisonic Surround-Sound Technology for Recording and Broadcast," published in the December 1983 issue of *R-e/p*, could you please be good enough to advise me of those recording companies who are producing analog albums in Ambisonic.

By the way, I found this issue in particular to be most interesting. To me it seemed to contain a variety of different articles, all very detailed and informative, which I found of sufficient value to devote my time to read.

Editor's Note: Nigel Branwell, president of Audio+Design/Calrec, manufacturer of various Ambisonic encoders, decoders and control units, has provided *R-e/p* with the following list of record companies releasing analog UHJ-encoded material:

Nimbus Records
c/o German News Company, Inc.
218 East 86th Street
New York, NY 10028
(212) 288-5500
Contact: Fred Hoeller

The Nimbus catalog contains a cross section of UHJ-encoded classical, jazz, bluegrass and country titles, the majority of which were recorded live using a Calrec Soundfield Microphone.

Associated Production Music (APM)
888 7th Avenue
New York, NY 10106
(212) 977-5680
Contact: Phil Spieler

And on the West Coast:

APM
6255 Sunset Boulevard
Hollywood, CA 90028
(213) 461-3211

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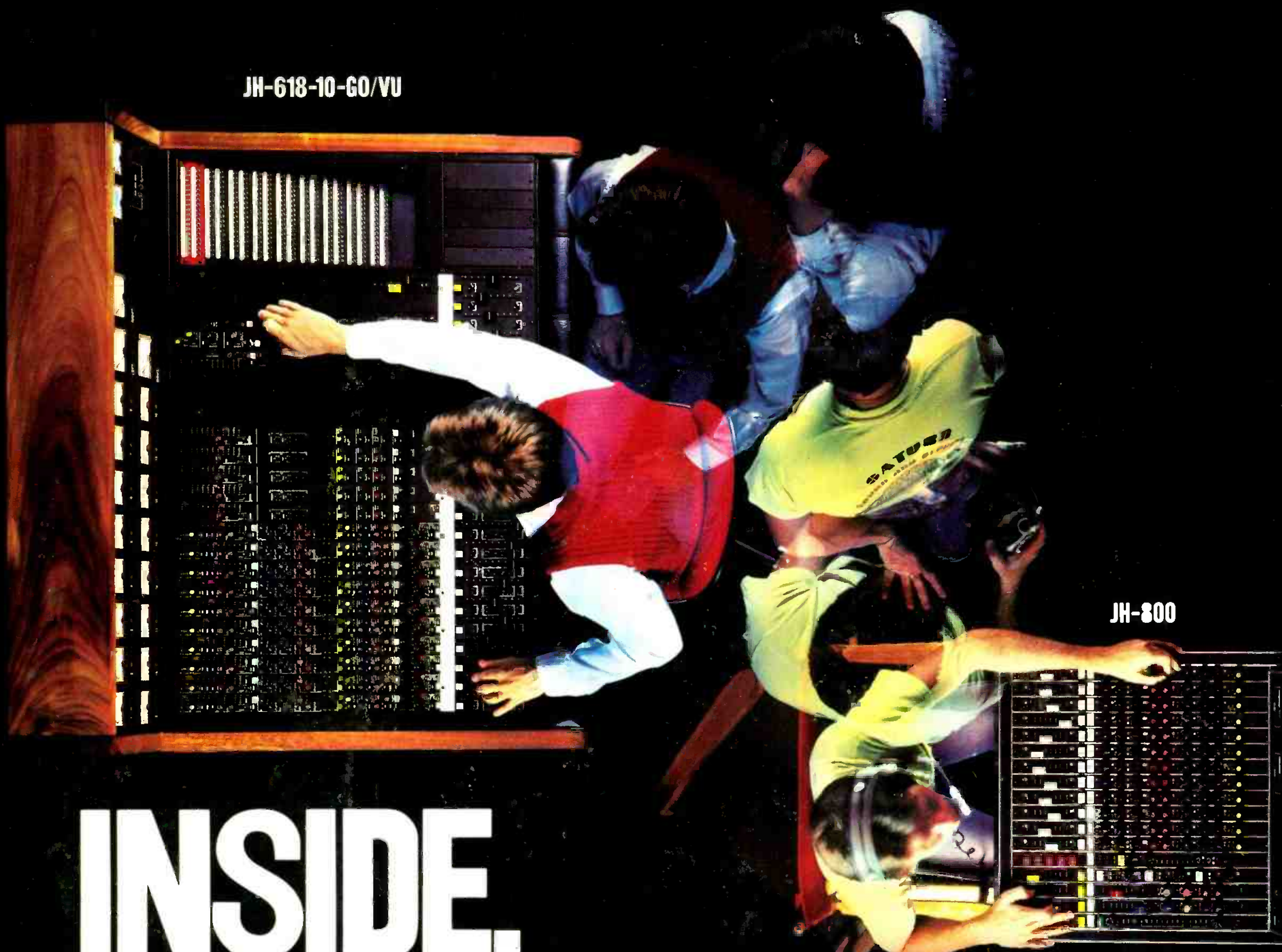

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For additional information circle #7

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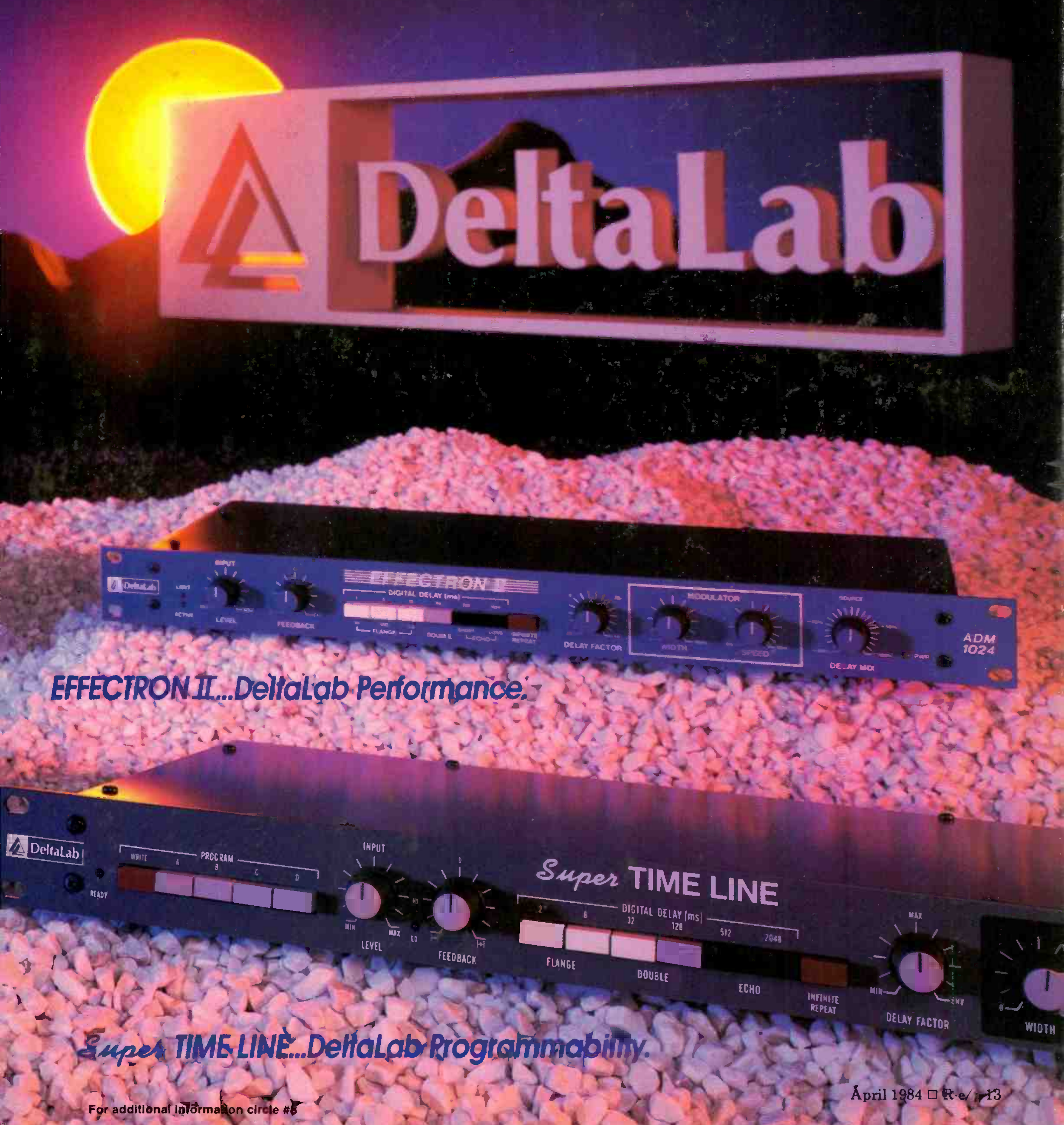
ECHOTRON™

The Echotron is a full bandwidth (16KHz at all delay settings), solid state digital delay loop. With a delay range from 256ms (¼ second) to 4096 (4 seconds) the Echotron is capable of producing very subtle to very spacey effects. Using the METRONOME feature and the DELAY FACTOR allows the musician to create bass lines or rhythm tracks. Hard and soft (roll off) feedback and synchronization for drum machines top off the versatile features of the Echotron. Only $699.00*



For additional information circle #8

LETTERS

— continued from page 10

The KPM music library selections available through APM include multi-track derived pop music that was mixed using the ADR Ambisonic Mastering System. □□□

REFERENCE LEVELS

from: **Phil Wells, chief engineer**
KPSI AM/FM
Palm Springs, California

In your December 1983 issue of *R-e/p*, I read with great interest and appreciation, John Roberts' iconoclastic article on level references, decibels, etc. After absorbing that, I read a Letter to the Editor which mentioned the confusion between "dbu" and "dbv" and their references of 0.775 and 1.0 volts.

In broadcasting we don't have that problem. To us broadcasters "dbμ" means "decibels with reference to one micro-volt," and is most commonly used when defining field strength contours for FM broadcast stations (60dbμ=1 millivolt per meter).

Also, for your amusement, we use "dbk" as "decibels with reference to one kilowatt" (an FM station with an effective radiated power of 100kw radiates 20dbk); dbi as "decibels with reference to isotropic"; and dbd as "decibels with reference to a dipole" (equalizing parameters in determining antenna gain). In the radio receiver bizz, they use dbf as "decibels with reference to a femtowatt," to determine receiver sensitivity in lieu of microvolts.

Personally, I've always appreciated "dbx" as "decibels with respect to the effectiveness of noise-reduction," and would recommend "dba" as "decibels with respect to anything." Though some perfectionists may scoff and say that this suggestion may prove too ambiguous, I ask you . . . can it get any worse? □□□

XLR POLARITY

from: **Randy Opela,**
marketing manager,
Switchcraft, Inc.

While we here at Switchcraft feel that the term "XLR" should be replaced immediately with "Q-G" (Switchcraft's trade name), *R-e/p's* questionnaire is very timely. [The questionnaire, which was sent to pro-audio manufacturers earlier this year, asked which pin of an XLR-3 the firm used as "hot"; results are to be found in John Roberts' "Audio Mythology" column elsewhere in this issue — *Editor.*] Having been a performing musician for 20 years (prior to joining Switchcraft) and also being involved with recording and sound reinforcement, I had bumped into the polarity question many more times than I care to remember.

I became aware of IEC 268 upon reading Ken Dibble's article in *Studio Sound*, February 1982. Since that time, I have received approximately two calls per month, asking if pin #2 or #3 was "hot."

Although corporately neutral, since we

. . . continued overleaf —

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Hold a seashell to your ear and hear nature's pure and natural sound. Compare it to the final sound produced by a Neve. You'll begin to understand why Neve is far and away the most respected name in audio mixing.

The Neve sound is so pure and natural, one might suspect that nature herself had a hand in the design. Perhaps.

Not every engineering achievement can be explained away. There are mysteries in nature, just as there are mysteries man-made.

Some suggest that Neve's unique Formant Spectrum Equalizers account for its unparalleled sound by taking into account the psycho-acoustic properties of voice and music while satisfying the critical demands of balance engineers. Others claim it's superior technical performance, novel circuitry, or high quality components.

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LETTERS

XLR Polarity . . . *continued* —

do business with both camps, I have privately advised our customers to use pin #2 as hot for any equipment of new design. Beyond that, I explained that there were standards (IEC 268 and BS 5428), and that all that could be done was to "match" their wiring scheme to that of the interfacing equipment.

Those customers of ours who are in the cable business indicate that they used to see no appreciable quantity differences (in unit volume, pin #2 hot versus pin #3 hot) prior to 1983, but that in the past year they are indeed seeing more and more requests for pin #2 hot versions. Most studios and broadcasters who I have questioned also indicate the same. Further, Peavey, Sunn, BiAmp, Audio Technica and Electro-Voice/Tapco are all using pin #2 as "hot." To my knowledge, Shure Bros., however, is still using pin #3 as "hot."

In short, all input and information I have received indicates a ground swell of support for the already established IEC 268 standard, and I plan to continue recommending compliance to this standard.

To further complicate your life (and possibly provide a topic for future articles), we at Switchcraft are now developing connectors and receptables for use as "hi-power speaker connectors." We are planning for this series to become another industry standard. I will keep *R-e/p* readers advised as to the status of this program. □□□

LUCAS/McFAUL — A Correction

from: **Bill Barton, general manager**
Lucas/McFaul, New York City

We read with interest the long article Lucas/McFaul, Inc. and the Warehouse Recording Studio [published in the December 1983 issue of *R-e/p*]. However, I found one important factual error in one of your studio layout drawings.

Our control room was built by Sierra Hidley Design, *not* by Eastlake. We're very please with both the Tom Hidley design and with the Sierra construction, so we want to give credit where it is due. □□□

EXPOSING AUDIO MYTHOLOGY

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

by John Roberts

This month I am pleased to present the results of the *R-e/p* XLR-3 Survey. As you will recall from my April 1983 column, there is more than one way to wire up a three-pin connector, and about the only thing anybody agrees on is grounding pin #1. Read on to see how your favorite company is doing it. In Part Two I will attempt to say something positive about the much maligned negative feedback.

How to Use This Survey

As can be seen from a quick glance at the accompanying list, there will be many opportunities for the system integrator to get his/her wires crossed. It is worth noting that for much balanced-in/balanced-out (in-line) gear there is absolutely *no* difference between calling pin #2 or pin #3 hot. We usually get into trouble when interfacing single-ended (SE) equipment with other SE (unbalanced) gear.

Thanks to the results of this survey, you will now know which machine to take your hammer to. Actually a curt note on the warranty card, something like "Please see IEC 268-12," should do wonders. But don't expect instant results — it isn't easy for a manufacturer to change over to pin #2 hot, and become incompatible with his entire installed base of equipment. However, to my mind at least, there is only one "correct" way to do it (Pin #2 hot), and just maybe someday everybody will do it the same. In the meanwhile, I would like to suggest to the manufacturers that aren't already doing so, that they note on the rear panel how they are wiring the connectors in a given unit. The hardest part for the system integrator isn't making the cables,

it's figuring out what cables are needed.

If I could make another observation about interfacing SE equipment, most of the problems are caused by SE wiring. Let me explain. Most professional SE equipment uses differential-input amplifiers, which will accept signals from both pin #2 and pin #3 similar to a balanced input. An SE signal at pin #2 will look *identical* to an SE signal at pin #3 of opposite polarity. Problems occur when using SE wiring and guessing wrong as to which pin has the signal.

(NB: While differential inputs are often referred to as "electronically balanced," there are subtle differences between differential and true balanced, regarding impedances to ground and ground-current flows. When properly configured, the differential amplifier is a slightly better match to unbalanced signal sources.)

If you don't see your favorite manufacturer on the survey list, it's because we hadn't received a completed questionnaire from them by press time; we'll do a catch up for the stragglers in a future column.

RESULTS OF XLR-3 SURVEY

Company	Pin 2/3 Hot?
AB Systems Design	2
ACO Pacific, Inc.	2
AKG Acoustics, Inc.	2
Amek Systems + Controls, Ltd.	2
Analog Digital	2
Aphex Systems, Ltd.	3
Auburn Sound Corp.	2
Audio+Design/Calrec	2
(Pin #3 being changed on some units.)	
AXE Artist X-Ponent Systems	3
Brooke Siren Systems	3
Bryston	3
Calrec Audio, Ltd.	2
Clear-Com Intercom Systems	3
Crown International	2
Datatronix, Inc.	3
David Hafler Company	2
DOD Electonics Corp.	3
Electro-Voice, Inc.	2
Fender Musical Instruments	2
Fostex	3
Gotham Audio Corp.	2
Henry Engineering	3
HM Electronics	2
Holmes Corp.	2
Inovonics, Inc.	3
Jensen Transformer	2
(Some old units, pin #3.)	
Klark-Teknik Electronics	3
Kudelski	2
Lectrosonics, Inc.	3
Lexicon, Inc.	2
MXR Innovations, Inc.	3
Neotek Corp.	3
Orban Associates, Inc.	3
Otari Corp.	3
Pacific Recorders	3
(Pin #2 to spec.)	
Quad Eight/Westrex	3
Quantum Audio Labs, Inc.	2
Ramko Research	3
RAMSA/Panasonic	3
Rane Corp.	2
Sescom, Inc.	2
Shure Brothers, Inc.	2
Simon Systems	3
Sony/MCI	3
(Pin #2 on mikes and mixers; choice for tape machines.)	
Soundcraft Electronics	3
Soundcraftsmen, Inc.	2
Soundworkshop	3
Speck Electronics	3
Studer Revox America, Inc.	2
TEAC Corp of America	3
(Mikes are pin #2 hot.)	
The Sound Steward	2
Tristech/Audio Envelope Systems	2
Turbosound, Inc.	2
Ursa Major	3
Valley People, Inc.	3
(Pin #2 per customer.)	
Westlake Audio	3
White Instruments	3

NEGATIVE FEEDBACK

It has recently become fashionable to blame negative feedback for a host of sonic ills. The argument is usually raised in the hi-fi power amp marketplace, but I think we should take a look at it because just about every audio path in your studio has circuitry that contains large amounts of negative feedback. If there is something wrong with NFB, I certainly would like to know about it.

If necessity is the mother of invention, then NFB can trace its parentage to the early telephone systems (Ma Bell?). Transmission of voice signals over even moderate distances required numerous booster amps to make up for transmission losses. Now, suppose a good amp for the day had ±1 dB frequency response, and 1%

. . . continued on page 25 —

QUANTEC SIX-PAGE INSERT FOLLOWS ON PAGES 19 thru 24
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For additional information circle #13

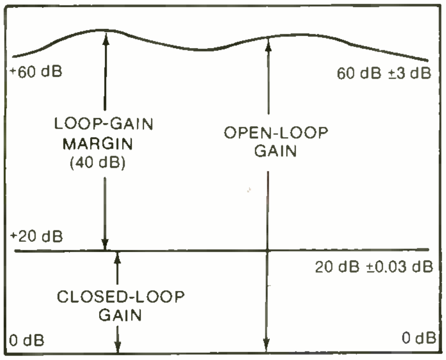

FIGURE 1

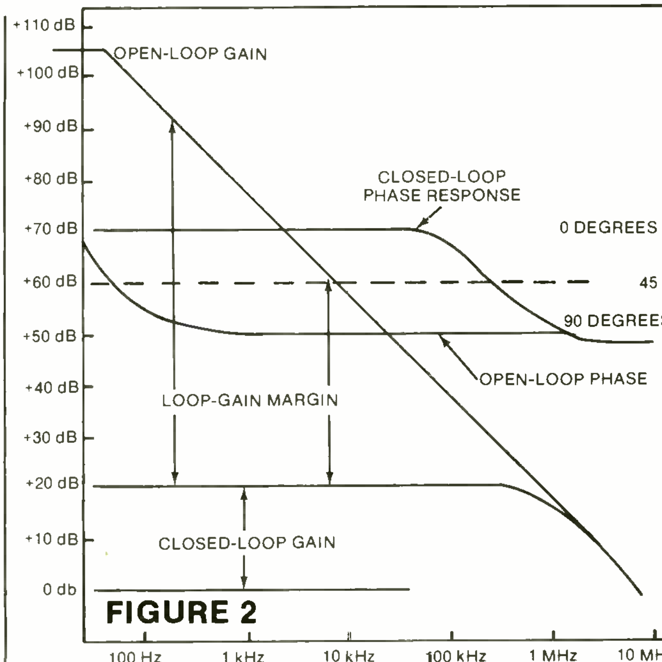

FIGURE 2

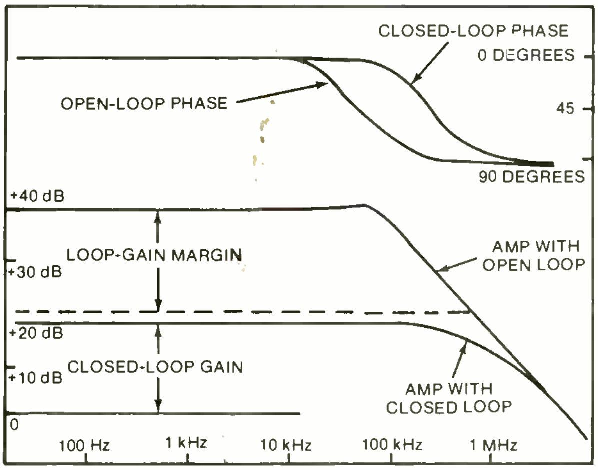

FIGURE 3

— continued from page 18 . . .

distortion. It's easy to see how passing a signal through even 10 such amplifiers could garble the signal beyond intelligibility.

Around 1920 or so, Harold Black, a telephone company engineer, invented negative feedback. By feeding back a portion of the output signal to an amplifier's opposing input, it is possible to greatly improve linearity (approaching the proverbial "straight wire with gain"). All of an amplifier's open loop non-linearities, such as distortion and non-flat frequency response, are reduced by the ratio between open-loop and closed-loop gain (loop-gain margin). For example, an amplifier with open loop gain of 60 dB (1,000x), a frequency response of ±3 dB, and 1% distortion, configured by negative feedback for a closed loop gain of 20 dB (10x), would have a loop-gain margin of 40 dB (100x) — see Figure 1. This compound amplifier would actually deliver a gain of 9.99x for a gain error of -0.009 dB. The frequency response would be ±0.03 dB, and distortion a respectable 0.01%.

As is usually the case with mother nature (father physics) there is no free lunch! In this case, however, the cost is moderate, since we trade gain for improved accuracy, and at audio frequencies gain is cheap. General purpose op-amps often begin with more than 100 dB of open-loop gain.

And Now for Something Negative . . .

Alas, there is a fly in the ointment. Amplifiers have a small but real time lag between when you tell the input of any amplifier to zig, and the output actually zigs. This delay is normally some small fraction of a microsecond, and certainly not audibly significant. However, something funny happens because of that negative feedback path. There will exist some frequency for which the input will already be zagging by the time the zig works its way through the amplifier and feedback network, and back to the input. This zig at the minus input happening at the same time there is a zag at the plus input is no longer negative feedback; Mr. Black's low-

WHY IT'S OTARI FOR MORE POST-PRODUCTION PROS.

Today's hot topic is audio post-production. At Otari, it was a hot topic years ago—and still is.

In 1979, we introduced the MTR-90® multi-track recorder which featured our new pinchrollerless tape drive system. It was quickly accepted as a superior performer under syncronizer control—with faster search and lock-up for quicker, easier sessions.

In 1980, we introduced the MTR-10/12® ¼-inch and ½-inch machines—the industry's first with microprocessor-controlled transports.

In 1981, the Series II MTR-90 provided even faster, more accurate tape handling, and was the first audio machine to offer a serial port for computer control of the transport functions.

Along the way, we've worked with every major synchronizer and video editor manufacturer to provide our customers with the interfaces they require. Synchronizer interface connectors are now standard equipment on all MTR Series recorders.

Our commitment to audio post-production continues.

In 1983, we adopted the IEC standard for center-track SMPTE/EBU time code recording on ¼-inch tape. We also developed a 1-inch C format audio layback recorder conversion kit for the MTR-90, and an RS232C Serial control port for the MTR-10/12.

Now, we're introducing the EC-400 Series, an advanced tape speed resolver capable of phase lock over a ±30% frequency range.

These are just a few of the ways we've delivered on our commitment to the audio post-production professional. To get the complete picture, contact your nearest authorized Otari dealer.

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The EC-401 Resolver: A precision speed controller with Widelok™—everything necessary to lock a tape machine's speed control track to any internal or external time base reference. Available as a compact, rack mount package, or for the MTR-10/12, as a plug-in module.

Technology you can Trust.

distortion amplifier is now a several-MHz oscillator. Since the telephone company was not yet interested in building wireless radio telephones, a fix was needed. Messrs Nyquist and Bode demonstrated that an amplifier would be stable as long as the product of the open-loop gain and feedback path attenuations added up to less than unity gain before the amplifier's time delays caused 180 degrees of phase shift.

The most common method to reduce the open-loop gain of a multistage amplifier is to place a capacitor across an intermediate gain stage. This causes the amplifier to take on the transfer function of an integrator — namely an amplitude response falling at 6 dB per octave with 90-degree phase shift from input to output. The power of negative feedback is such that even with an open-loop response that looks more like a scratch filter, the compound amplifier will still deliver ruler flat response over a wide range of frequenies.

Figure 2 shows a plot of the open-loop gain of such a stabilized amplifier. As car be seen, at very high frequencies there is inadequate loop gain margin to hold the closed-loop gain, and the response falls off. Also included in Figure 2 is a plot of open-and closed-loop phase response. As with the other open-loop non-linearities, the 90-degree phase shift has little affect on the closed-loop response, as long as there is sufficient loop-gain margin.

Low Negative Feedback?

There are some designers who prefer the open-loop response plotted in Figure 3. While a constant loop-gain margin over the audio band will make calculation easier, I do not expect much, if any, audible difference. The gain error will be larger but flat, and therefore not audibly significant. The distortion will also be larger, but again (within reason) insignificant. Both amplifiers will have similar transient response (frequencies above F1), with the low feedback design having less phase shift below F1. As the phase shift of amplifier #2 is already respectably small in the audio band (less than 1 degree below 20 kHz), I find the merit of low feedback design to be elusive. Again, I would not expect to hear a difference between competent executions of these two design approaches using similar components.

While not perfect, negative feedback is an extremely powerful technique, whose only flaws I can find show up when you don't use (or have) enough of it. Its widespread use has been instrumental in attaining the high levels of performance that we take for granted in today's equipment.

In a future column I will address the relationship of amplifier slew rate to performance, sonic or otherwise.

Note: There are applications where *positive* feedback is useful. For example, should you want to build a signal generator, positive feedback can be used to sustain oscillation in a tuned circuit. Another common use is to bootstrap or increase the input impedance of a circuit. See, even positive feedback can be good! □□□

TRAVELS WITH THE EDITOR

Report on SPARS "Digital Audio on Trial" Seminar, University of Miami, March 8 thru 10, 1984

by Mel Lambert

With worldwide sales of Compact Disc players and software currently riding high, and the pro-audio industry at last catching a glimmer of financial hope at the end of the recessionary tunnel, the technology of the Eighties looks set to be Digital. Sony is reporting healthy sales of the PCM-3324 digital 24-track on half-inch, while Mitsubishi also is fairing well with its X-800 32-track/one-inch transport (now gracing the control rooms of Lion Share, Fantasy, and United Western on the West Coast, plus New York's Audioforce equipment rental company). And all this against a background of routine digital mastering to Sony PCM-1610 and PCM-F1/710 (plus similar EIA-J-format 14- and 16-bit processors), JVC DAS-90/900, Mitsubishi X-80, 3M DMS, and Soundstream machines. (Both Sony and Studer recently unveiled DASH-format stereo digital machines — see accompanying sidebar for further details.)

If nothing else, the professional audio community — and this should be taken to include film post-production, broadcast, audio-for-video, and allied fields, as well as the more visible record-company sessions — has now woken up to the fact that digital is, without doubt, the wave of the future.

Against such a background of growing interest in things binary, the society of Professional Audio Recording Studios is to be applauded for having organized a particularly interesting seminar program entitled "Digital Audio on Trial," and held in early March in cooperation with the University of Miami at the latter's Gusman Hall in Coral Gables, Florida. Even though members of the SPARS executive committee were expressing a certain amount of disappointment at the turnout — slightly less than 120 individuals registered for the three-day conference - to this observer at least the quality of the attendance, and the high calibre of the ensuing discussions before, during and after each day's spirited sessions, more than made up for the lack of numbers.

Which is not to say that subsequent seminar programs could not benefit from a more healthy attendance. As will be seen from the following report on the conference proceedings, practically all digital options are still up for grabs, and without a continuing debate — including a critical appraisal of the pros and cons of each system approach — our industry may well end up with a digital standard that proves to be insufficiently flexible to suit its expanding needs.

Kicking off the proceedings on Thursday night, Ken Pohlman, director of the University of Miami's Music Engineering Technology Program, provided a necessarily brief overview of digital recording technology in his "Introduction to Digital Audio" seminar. Pohlmann's talk covered the problems of lowpass filtering to provide a "brick-wall" response at half the sampling frequency; sample/hold circuitry; the complexities of various analog-to-digital conversion process, including the addition of dither signals; writing binary data to the storage medium (with a particularly succinct description of error detection and correction schemes for fixed-head and helical-scan/VCR transports); plus subsequent data recovery and D-to-A processing.

Friday morning began with a lively debate that addressed the premise: "Digital Audio on Trial: If it's so good, why is it so bad?" Moderated by Michael Tapes of Sound Workshop, the discussion took the form of a reply to Tape's conviction that the Compact Disc sounds inferior to an equivalent analog release. It should be pointed out, though, that Tapes deliberately chose to play the role of Devil's Advocate, and went to great lengths to stress his belief that it was *vital* that digital technology remains under critical scrutiny from our industry. Otherwise, he offered, we run the grave risk of accepting an inferior technology too soon, without coming to grips with all the potential alternatives. So, having stated candidly that he and his company have both financial and technical reasons for wanting digital to succeed, Tapes offered that, to his ears at least, "digitally recorded music is not emotionally pleasing."

Quickly coming to the defense of digital, panelist Bruce Botnick of Digital Magnetics, Hollywood, felt that the current debate regarding sonic differences between analog and digital can be likened to changes the recording industry experienced during the transition from tube to solid-state consoles in the Sixties. "The change in sound was unbelievable," he remembers. "The technology improved, and we got used to it. Also, I recall the introduction of Dolby noise reduction. We had to learn to 'listen through' the system, and adjust EQ and echo levels to suit the new medium."

Roger Nichols of Soundworks Studios, New York City, agreed that we all had to go through a learning curve when coming to terms with digital. "We have to alter our recording techniques now that there is no need to make up for deficiencies in the analog recording process," he said. "Differences we hear on digital pressings *may* be due to a certain non-mastery of the medium; there is a definite learning curve for engineers."

"Digital can produce a very *different* sound from analog," Botnick added. "For example, the first 50 CDs from CBS were made from quarter-inch EQ copies, or a second-generation copy of that tape. Other CDs come from all-digital sessions, so it's hardly surprising that there are many variables in the process."

"We need to compare CD or vinyl releases with live source material to make a valid comparison of the sound quality," Nichols offered. "And reviews in the hi-fi press are often based on what the reviewer *thought* it should sound like!" (As an aside, Nichols recalled that he had made some comparisons between digital recordings on both 3M DMS and Mitsubishi X-800 32-track machines, and a simultaneous live source, and could detect *no* differences between any of them.)

Addressing Tapes' comment that some reviews in the hi-fi press had claimed CD sounds far better than vinyl, which to Tapes was obviously *not* the case, Nichols stressed that we need to check the source material used for CD mastering. Referring to points similar to those made in his article published in the December '83 issue of *R-e/p*, Nichols made a plea for ensuring that the highest quality source material be used — *not* a multiple-generation analog copy that obviously will sound inferior to the tape used for vinyl mastering.

John Eargle of JBL and JME Associates offered that analog tape and vinyl "grew up together," and that "we may need to modify the message to fit the medium, because we now know exactly what the result will be in the home environment." We are faced with two problems, he says: analog tape saturation that causes HF self-erasure, and a resultant softer-sounding high-end; and vinyl pressing

. . . continued on page 33 —

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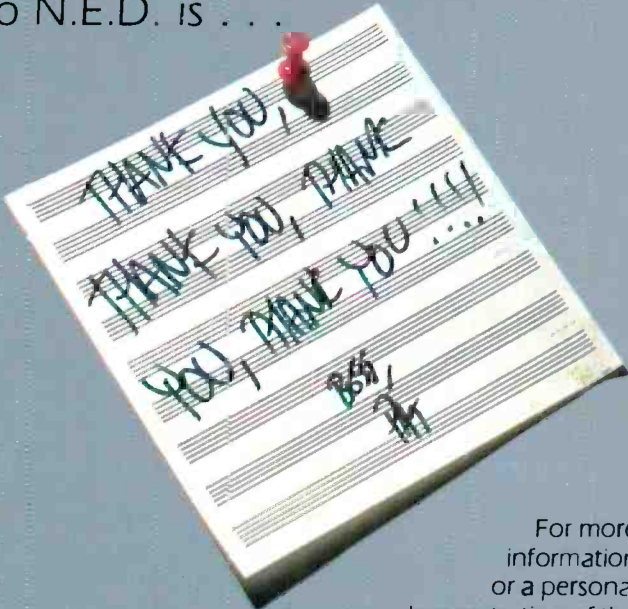

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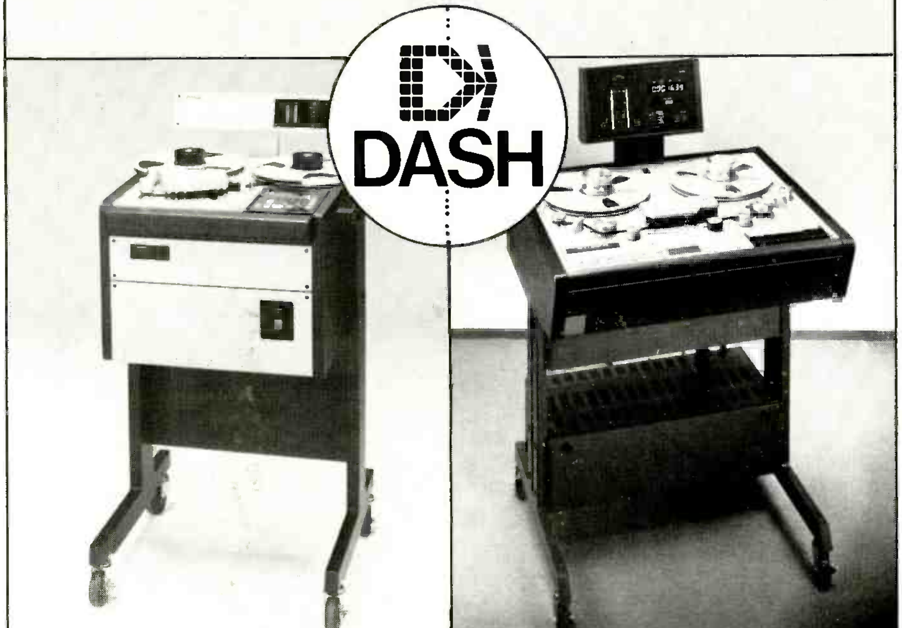
quality, which often means that a cutting engineer has to cut the track a little hotter to overcome surface noise, but then runs out of headroom. Mastering a CD release from digital presented no such difficulties, and also posed few problems during domestic playback.

Turning to practical differences between analog and digital, Eargle offered that if you compare a solo piano replayed from CD and vinyl, the digital produces "better time-base stability, has no wow and flutter, and produces reduced groove echo." However, such differences often need to be pointed out to people before they notice them, and then maybe use such advantages as subjective criteria.

Responding to audience comments about the enhanced durability and lack of wear offered by the Compact Disc, Tapes reiterated his hope that the present seminar address itself more to pro-audio considerations, in particular the role of digital in the recording studio — and maybe that CD tapes do not necessarily need to be mastered digitally. Botnick was the first to respond to the challenge, by commenting that he prefers the subtlety offered by digital recording, in particular the enhanced stability of material panned between a pair of tracks. Nichols stressed the operational advantages offered by dig-

STOP PRESS — LATEST DIGITAL NEWS

Sony and Studer Unveil DASH-Format Stereo Digital Machines; Mitsubishi X-800 32-Track Purchased by The Burbank Studios



At the recent AES Convention in Paris during late March, both Sony and Studer demonstrated newly developed DASH-format stereo machines. Both the Sony PCM-3102 and the Studer PCM transport record a total of eight digital tracks (four per audio channel) and four auxiliary tracks (SMPTE, control, plus two analog audio channels) across quarter-inch tape running at 7½ IPS. Sampling frequency is 48 kHz (selectable to 44.1 kHz at 6.89 IPS). Up to two hours of recording time is available with 10.5-inch reels. Both machines support full editing capabilities — either electronic, or with "conventional" razor-blade techniques. While the Studer digital machine on show in Paris was a prototype — laid out as a transport with a separate package housing the A-to-D and D-to-A electronics, thus enabling console or table-top versions — Sony reports that eight PCM-3102 machines have been sold to NHK, the Japanese radio and TV organization. To meet the specific requirements of broadcast facilities, the NHK digital machines have a monitoring overbridge that houses cue amplifiers and loudspeakers. Although Sony has not to set a final price on the PCM-3102, according to Studer its digital machine, which the company plans to make available within a year, has a target price of "around $20,000."

Also under development at Studer is a DASH multitrack that will use thin-film head technology to record 48 tracks across half-inch tape running at 30 IPS (the DASH double-density format). The new multitrack will provide 32 channels of digital audio, plus eight DASH-encoded tracks for "improved reliability of the multichannel format," and an additional eight DASH-encoded tracks carrying auxiliary digital audio data for use as spare channels, or for extended dynamic range and other applications. (Incidentally, Studer considers that 24 digital tracks is "too limited a number," but that 48 tracks "are not required in today's applications" — hence the choice of this new "Cross-Protected DASH" 32-track recording format, with 16 auxiliary digital channels, and four analog, SMPTE and control tracks.)

• The Burbank Studios will be taking delivery of a Mitsubishi X-800 32-track on one-inch digital machine, for use in post-production for film and television. The new multitrack is expected to be installed during May in Scoring Stage #1, but reportedly will be utilized at various locations around the TBS complex. □□□

PJ5055

ital, referring to his sessions with Steely Dan and Donald Fagen. "We do a *lot* of overdubs and punch-ins," he said. "Digital is very tight in and out of a punch — far more than analog. Also, back-up/safety reels are easier to create with a digital multitrack; we went up to 12 generations down on the 3M DMS with overdub copies, and could detect *no* differences with the first generation tapes." Botnick added that analog also loses substantial high-end over time, which was not the case with digital.

Winding up the panel discussion, Eargle pointed out that although the "semi-professional" Sony F1 achieves 16-bit capability by "stealing" two bits from the error-detection data, it still sounds very good due to sophisticated error concealment. And, addressing the problem of compatibility between systems, Nichols offered that with multitracks there was little problem of swapping tapes from machine to machine within the same brand, but it should be remembered that analog tapes seldom sound the same on different machines. Mastering compatibility was another matter, however, since stereo master tapes move around studios a lot more than multitrack reels.

The second session of the day, "The Recording Studio and the Compact Disc," was moderated by Bob Ludwig of Masterdisk, New York; panelists included Roger Nichols, and Bill Foster of the London-based Tape One analog cutting and CD mastering facility. Responding to Ludwig's assertion that, because digital is still in its infancy, our industry needs "qualified people in charge of CD mastering and pressing plants," Nichols agreed that during preparation of digital tapes for CD mastering you need to ask a *lot* of questions: What format? What sampling fequency? What recording level? etc, etc. Since practically all of the points covered in Nichols' useful discussion are detailed in his article, "Preparing Digital Master Tapes for Compact Disc Mastering," to be found elsewhere in this issue, I won't duplicate them here, but would simply reiterate his plea that to ensure correct transfer of digital information to Compact Disc — and not simply leave it up to the CD laser cutting plant — then it really does behove a producer or engineer to find out more about the various technical parameters of the 1610-encoded U-Matic master tape (or whatever format is acceptable to the plant).

Turning to the future of the all-digital studio, Foster mentioned that his facility was currently awaiting delivery of not one but *two* Neve DSP cutting and CD mastering consoles. At present Tape One prepared CD master tapes from original digital masters by dubbing and re-adjusting levels and EQ in the analog domain, or through a DAE-1100 editing system; installation of the pair of Neve consoles will enable processing to be carried out totally in the digital domain. (A prototype Studer digital Sampling Frequency Converter is also available for transcoding non-1610-encoded tapes to the format required by most European and Japanese CD mastering facilities.)

The first digital board, scheduled for delivery in April of this year, will serve as a CD Mastering Console, and is basically a stereo-in/stereo-out configuration equipped with four-band digital parametric EQ, low- and highpass filters, compressor-limiters, and +10 dB of available gain. The console will be set up for 44.1 kHz, PCM-1610 video-format input and output (or analog-in via the 1610's A/D), and will enable CD master tapes to be re-equalized and level adjusted in the digital domain prior to dispatch to the cutting plant.

The DSP Disk Mastering Desk, due for delivery this summer, will feature the same EQ, filtering, gain riding, and compression capabilities as the smaller board, but will be of a stereo-in/four-channel-out configuration (one output pair for discrete preview, with adjustable digital-delay offsets, and another pair of analog disk-cutter signals), and will operate at 48 kHz digital in/out. As well as accommodating conventional analog in/out signals, the new board will be equipped with five additional stereo digital outputs set up at various points in the signal path, including pre- and post-EQ, to enable digital copies to be struck during analog disk mastering; a full AES digital interface also will be incorporated. Unlike the smaller console, which will utilized the PCM-1610's LED peak meters for level display, the more complex disk will incorporate bargraph displays, plus a full set of Neumann lightbar, VU, and phase-correlation metering. Full remote control of Tape One's Neumann VMS80 analog cutting lathe will also be included. To enable identical functions to be set up for recuts, a maximum of 20 "snapshots" of the digital control surface can be stored on 5¼-inch floppy diskettes.

Having now prepared some 100 PCM-1610 tapes for CD mastering — Tape One also expects to take delivery of a Philips PQ Subcode Editor in the very near future — Foster had some interesting horror stories to tell about the sometimes appalling quality of tapes offered for CD release. By way of an example, he recalled that the digital master for Blondie's *Parallel Lines* album was 6 dB down in level from peak, had been dubbed off-azimuth from the analog master, and even had a fade out at the end of one track that suddenly muted half-way down the fade!

Responding to a question from the audience, "Why doesn't the record company, producer, and/or artist attend the mastering stage?", Ludwig offered that the producer may not even *know* a CD is scheduled for release, much less be invited along for the remastering session. It would be useful, he said, if the producer could attend the session, because there are often artistic reasons why, for example, the timing of sides originally had been chosen for vinyl release, and which with the CD's longer playing time may need to be adjusted. On a more practical level, Foster offered that when a mastering facility is presented with an original master tape — either analog or digital — "you should ask what EQ and level adjustments were made during the original analog cut, so that you can replicate them for the CD release." For some strange reason, he recalled, possibly because it had been mastered from a non-edited tape, the Compact Disc release of

Phil Collins' *Face Value* album was longer than the vinyl version!

The last session on Friday, entitled "Digital for Dollars," and moderated by Hamilton Brosious of Audiotechniques, New York, addressed the financial aspects of a studio making the move to digital recording and mastering. Record Plant's Chris Stone recalled that his LA studio made the transition five years ago, but that within 18 months he had returned the digital multitrack to the manufacturer. "Because of an industry downturn, the timing of doing digital wasn't good," he recalled. "There was no CD then; the advantage for vinyl release was not sufficient to offset the cost involved. We needed to charge an additional $25 per hour over a 40-hour week to just break even — record companies simply wouldn't pay the difference.

"The key to success is rental: record companies will pay for renting a digital machine what they wouldn't pay as part of the hourly studio cost. Our new 3324 is costing $4,000 per month to lease. The only smart way to pay for it is to rent it back to the client."

Record Plant is seeing a lot more business from film companies interested in using digital for scoring sessions. Stone related that during dates for *Startrek III: The Search for Spock*, the musical score was recorded simultaneously on analog and digital multitrack, and also six-track mag — the digital sounded better than mag, he says. I also discovered that Stone and some industry colleagues recently completed a movie project entitled *Digital Dream*, for which the soundtrack was recorded entirely digitally; response from the Hollywood film community had been very strong, he conceded.

"Between three-and-a-half to four years ago when the recession bit," recalled Murray Allen of Chicago's Universal Recording, "we needed to think long-term. How do we increase sales? We could either put up our prices, or go for an increased share of market." Having decided to opt for the 3M DMS, Allen said that he found low-cost money by looking beyond the banks to investment groups and leasing companies. "We were lucky," he acknowledged, "because it was three years before the Sony and Mitsubishi systems appeared, and we still had a one-year lead time until other studios got them."

Allen also decided to promote and merchandise his studio's digital advantage. Hitting on the tag "The Tape Forgets," referring to the drop-off in analog quality over time, Universal gradually picked up an increasing amount of business from advertising clients. To recover his costs, Allen put up his session rate for digital by $50 per hour.

And there were additional advantages for advertising clients to be gained from digital. Apart from the creative edge of having 32 rather than 24 available tracks, for a jingle session that requires different vocal lines be laid over common orchestral backing tracks, multiple digital copies could be made without a loss of quality.

Summarizing his financial advice to prospective purchasers of digital hardware, Allen offered that a facility should work closely with a competent CPA and, if the cost is less than 5% of your projected operating budget for the next five years, then do it — if you cannot make up the additional 5% through increased business, then you might very well be in the wrong industry!

Readily admitting that he has yet to be sold on the idea of going digital, Sigma Sound president, Joe Tarsia, felt that he would find it difficult to pay for a digital machine if all he could charge was $25 per hour on top of his normal room rates. Sigma, he says, "is in the working-class business of making commercial records," and studio sessions at his New York facility are split roughly one-third each between advertising work, remixing outside tapes, and recording new music. Only the latter 33% of business is appropriate to digital, he offers, and that amount of work simply doesn't warrant the additional costs involved. "And from checking out the recent Grammy Awards, I notice that only the Police album involved digital — and then only for mixdown. So, the question I ask myself, is: 'How important is digital?'"

Brosious, whose New York rental company recently started to offer a selection of digital machines for hire, made reference to a recent survey of 18 East-coast studios, which indicated that 85% of them would be purchasing a digital two-track, while only one or two facilities were currently in possession of, or planning to buy, a digital multitrack. "A digital mastering machine will be a necessity by the end of '85," he predicted.

From the audience, Mack Emerman of

Criteria Studios, Miami, related that he currently owns a pair of Mitsubishi X-80 stereo decks, and that "pretty soon analog stereo half-inch will become secondary to digital. Since taking delivery of a second X-80, my half-inch machines are sitting in the studio corridor.

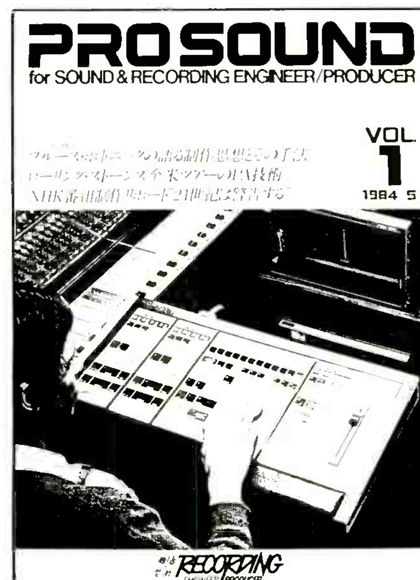
"Criteria has also used X-800 and PCM-3324 multitracks on sessions, and we find the sound to be significantly better than analog — a quantum leap in quality. Because of the ease of line-up, in a multiple-room studio such as ours, a digital multitrack makes a lot of sense."

Saturday's first session, "Standardization: Is it time?", moderated by Ken Pohlmann, took the form of a possibly inevitable comparison between digital formats. Curtis Chan of Sony provided an overview of Digital Audio Stationary Head (DASH), currently being promoted by Sony, MCI, Studer, and Matsushita. DASH supports two sampling rates (48 kHz, and 44.1 kHz at proportionally slower running speeds); three speed/track configurations (Fast/30 IPS with one data channel per track, Medium/15 IPS with two, and slow/7.5 IPS with four); four auxiliary analog tracks (SMPTE timecode, control track, and two audio channels); and between two- and 48-track configurations. Chan offered that such a format provided "the most powerful error correction available today," and supported both electronic and razor-blade editing. This statement he attempted to quantify by adding that DASH will handle "random correction of three [damaged] data words, perfect correction of a 9,640-bit burst error (0.52 cm), good concealment of a 33,982-bit burst error (2.23 cm), and concealment of an 88,232-bit burst error (5.51 cm)." And also that error correction was enhanced at slow speed by doubling the number of encode/decode channels — in some cases, he pointed out, even if one track is lost through tape damage, the other tracks' error correction is undisturbed.

Lance Korthals of dbx ran down the principles of Companded Predictive Delta Modulation which, through its 640 kHz sampling and "one-bit" (increase/decrease) operation, offered a 110-dB dynamic range (equivalent, he pointed out, to a 19-bit PCM system). For CD mastering the dbx system provided a significant headroom advantage over PCM, he said, which might only be capable of handling some 90 dB — very close to the dynamic-range capability of the Compact Disk.

Addressing the question: "Is it time to standardize?" Korthals offered that it probably wasn't, since we still are faced by several alternative systems. Also, CPDM is currently being made available to other tape-machine manufacturers, so that with time, if only because of its wide-scale availability, the dbx-derived format may form a standard.

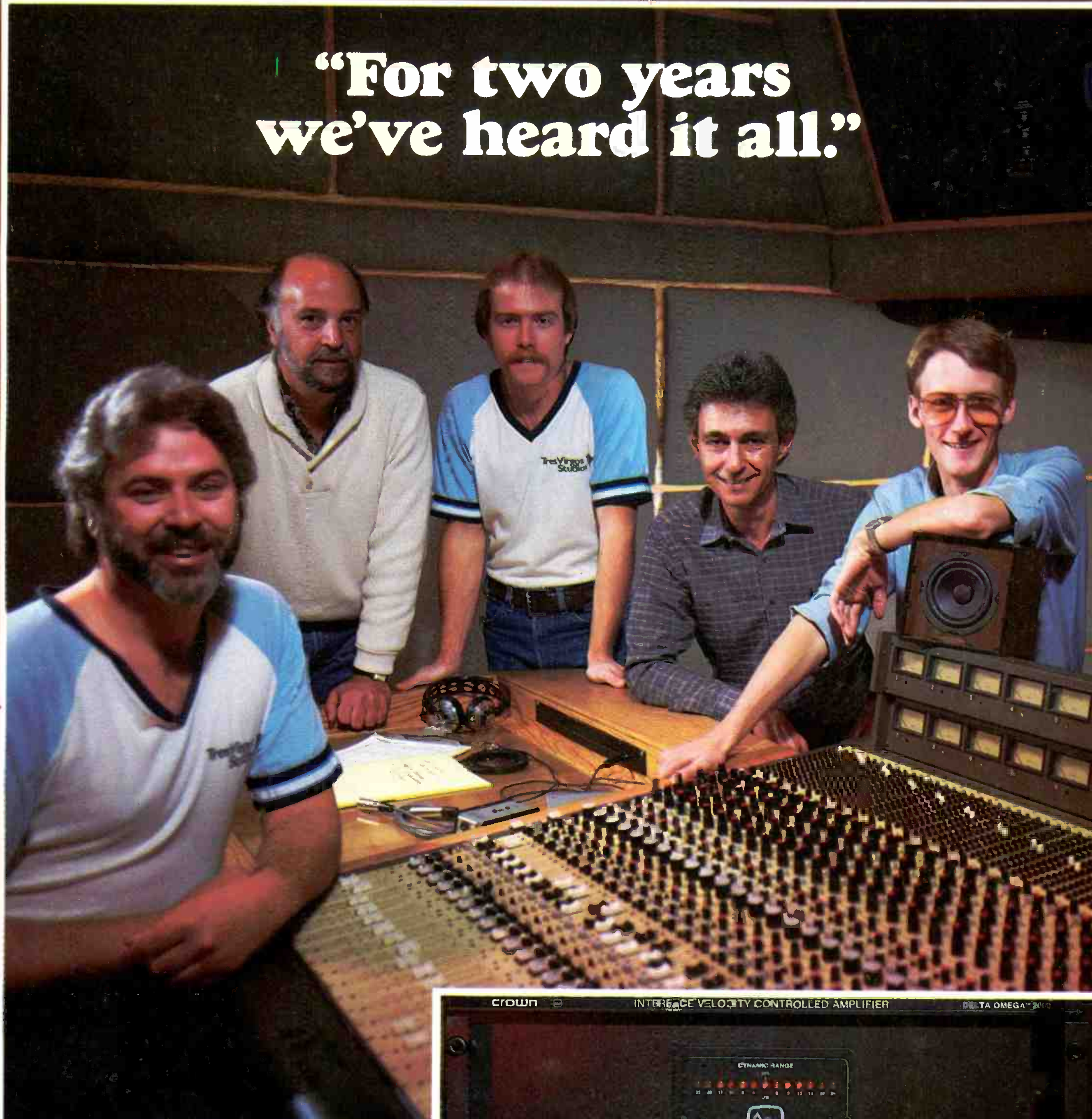
Replying to Chan's possibly rhetorical



NEW MAGAZINE FOR THE JAPANESE PRO-AUDIO INDUSTRY

Reproduced above is the front cover of a new publication aimed at the Japanese live-performance, recording and broadcast industries. Through a special licensing agreement with *Recording Engineer-Producer*, the new bimonthly magazine, titled *Prosound*, will include Japanese translations of articles that appeared originally in *R-e/p*. The first issue featured an interview with Bruce Botnick from the February 1983 issue, and an article describing Clair Brothers' sound system for the Rolling Stones Tour, which was included in our December 1981 issue.

The magazine's staff is headed up by publisher Isao Harada, editor Fumitaka Morita, production editor Yoshiji Tezuka, technical editor Susumu Nakamura, and contributing editor Larry Ishikawa. □□□



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question regarding why Mitsubishi hadn't joined the DASH caucus, Tore Nordahl of Digital Entertainment Corporation, which handles US sales and marketing for Mitsubishi digital products, offered that DASH was not strong enough in terms of error correction, and that 24 tracks was insufficient to satisfy the demands of today's recording sessions. Nordahl's criteria for standardization, he said, was three-fold: firstly, "Has CD arrive" — Yes; secondly, "Are there professional formats available" — again, Yes; and thirdly, "Will there be new pro-audio digital audio systems available soon?" — No. So, given these three answers, the time for digital standardization has arrived.

Why should a potential user select Mitsubishi digital? Not only does the X-800 offer 32 available tracks on one-inch tape, he said, the format's error correction is very robust. Being based on a pair of parity tracks per eight data tracks/channels, it is possible for an *entire* data track to be lost through tape damage, and the audio information reconstructed from the parity tracks, Nordahl claimed. Also, data synchronization can be derived from any one of the 32 digital tracks — not just the single control track utilized by the DASH format.

Robert Molstad of 3M's Data Cartridge Department Laboratory pointed out that, for certain practical reasons, the company no longer manufactures its Digital Mastering system. However, given the DMS pedigree, and the Seventies technology upon which it was based, there is still room for improvement in the system.

Running down the various parameters that need to be agreed upon for a workable digital standard — including mechanical compatibility, track layout and error correction schemes, plus tape magnetization techniques — he conceded that "If a standard is needed, then DASH fits the bill. However, it will probably be out of date soon." (It would appear that the good folk in Minneapolis are playing some cards very close to their corporate chests; time will tell if a MkII DMS appears on the market, and what digital format and storage medium it will adopt.)

Dealing with the philosophy of digital standardization, Almon Clegg of Matsushita offered that such a standard has to be voluntary, and also cost-beneficial for the user. "A manufacturer needs to look at the size of the potential market," he offered, "to decide if it can afford to innovate." As a result, "making such systems available at sufficiently low cost to the user may not be possible."

3M's Molstad pointed out that, for example, the development of certain custom VLSI chips may not be economical, given the size of the potential pro-audio market. Rather, he said, perhaps we should look at other storage mediums, including systems and formats already developed for the computer industry. Maybe the future lies in the various erasable, magneto-optical disks capable of storing up to five gigabytes of information, and which could hold up to 20, 30-minute digital audio tracks, he predicted. (Other storage systems that may bear closer scrutiny include Sony's DRAW — Digital Read After Write — system based on Compact-Disc technology, and Systex 300 currently under evaluation by Gotham Audio. The latter is said to be capable of holding 25 minutes of stereo digital audio on Winchester-type disk drives, with very fast access times.)

The conference's final session, "CD or not CD, was that the question?", moderated by Murray Allen, addressed the future implications of consumer and pro-audio digital. "We all have to be comfortable with our lack of knowledge about digital," said panelist Chris Stone. "The future may lie with DRAW for the Compact Disc. One application [of DRAW] could be to speed up sound effects selection and preparation in the film industry. Paramount, for example, spends many millions of dollars each year copying master SFX tapes for use on post-production dubbing stages." And random-access systems would be of great interest to the broadcast industry, he predicted, to replace NAB cartridge players, and for time-shifting news bulletins and announcements.

Expressing a note of caution, English journalist and engineer, Richard Elen, offered that "Now it might be possible to record a sound in the studio that can *only* be heard on a CD in the home." For example, low-level sounds, such as air conditioning, fluorescent lighting, and musicians moving sheet music, might go unnoticed in the control room — "either because there was too much happening during the session for the engineer to notice such subtle noises, or because the monitoring isn't as good as a domestic system." To explain his last remark, Elen hazarded that control-room monitors usually are designed for dependability, and high power handling capabilities; as a result, they may lack the degree of sonic detail possible from smaller, home loudspeaker systems.

By way of an example of the kind of odd sound that *should* have been noticed during the session, but which nevertheless found their way onto Compact Disc, the audience was treated to a digital tape copied from some rather "suspect" CD releases. During a quiet classical passage, the sound of a dirty jack plug was clearly audible — obviously the engineer involved in preparing the CD master was unaware of its effect — while from listening in mono to a vocal passage of "Stairway to Heaven" from the CD release of *Led Zeppelin IV*, it was blatantly clear that the head azimuth had been way off during the transfer from analog.

Consultant and hi-fi writer Len Feldman explained that, from conversations with consumers, the primary complaint against CD seems to be its lack of playing time — we were promised 70-odd minutes of digital audio, but most releases these days seem to run to not much more than 30. Although there are in excess of 1,000 CDs now on the market — a figure expected to exceed 3,000 by year's end — the quality of Compact Disc release was still being raised, Feldman related. And, as far as he could establish, the problem did *not* lie with sampling frequencies or bit rates, but rather with the software; early CD realeases did not sound very good, although now the quality of source material and manufacturing has got a lot better, he offered.

. . . *continued overleaf* —

Quoting Bob Carver's work on the sonic differences between CD and vinyl releases of the same title (and hopefully taken from identical master tapes), Feldman described the way in which Craver had compared the L+R and L–R signals of such software, and discovered that consistently over several dozen CDs there was 3 to 4 dB less difference signal than on the comparable vinyl offering. And that such a difference might explain the lack of stereo depth and perspective certain people claim they can detect on CD.

Stone ended the proceedings by outlining the recent SPARS labeling code for vinyl and CD releases. The three-letter code will identify, with either an "A" for analog or "D" for digital, the recording, mixing, and mastering medium. In January of this year Polygram agreed to adopt and implement the SPARS Code, Stone reported, and it appears that other record companies will follow suit before long.

During subsequent audience discussions, it transpired that, according to producer Karl Richardson, the next Bee Gees' album due for completion this summer, will be released simultaneously on vinyl and CD — with the Compact Disc, because of its longer running time, containing two additional tracks not to be found on the vinyl release.

All in all, the SPARS Digital Audio Seminar was considered a great success by practically everyone involved. A greater emphasis might have been placed usefully on information of greater relevance the pro-audio industry, including hands-on experience in the studio with various digital systems and formats, although the debate was sufficiently lively to satisfy most people's avid curiosity.

If nothing else, from the quality of the ensuing discussions, it would appear that there are still a multitude of questions to be addressed. Which, given the number of exciting options available to our industry, may not be a bad thing. ■■■

THE DIGITAL RECORDING FUTURE

Preparing Digital Master Tapes for Compact Disc Mastering

by Roger Nichols

This article will expand on the requirements of a Compact Disc Master Tape, including a step-by-step procedure necessary to produce a ¾-inch videocassette containing the final edited digital audio, SMPTE timecode, and PQ-Subcode necessary for the CD production plant. [Readers new to the CD production process might also like to refer to the author's article, "Potential Pitfalls in Mastering for the Compact Disc," published in the December 1983 issue — *Ed.*]

It is assumed that you now have in your possession an ORIGINAL master tape of the album, *not* a copy, and are ready to produce the tape necessary for Compact Disc production. The most common format at this time for mastering is the Sony PCM-1610 which utilizes the ¾ inch U-Matic videotape and recorder. The use of other formats by CD manufacturers will be discussed later. For the sake of clarity, I will stick to the PCM-1610 for the major portion of this article.

Initial Preparation Stages

If the master tape you are starting with is already PCM-1610 format, then you're one step ahead. If not, then it must be transferred. The two methods of converting from one format to another are through a digital sample rate and format convertor, or through analog connections between the two machines. The analog method is used most often, because it's the easiest, and is the same way you have always copied a tape from one machine to another. The digital-to-digital method will be persued in a few minutes, depending on how fast you read.

The level at which you transfer the tape is very important. Adjust the levels until the hottest section of the album leaves *exactly* 2 dB of headroom as indicated by the PCM-1610 level meters. (Peak-hold modifications can be made easily on the PCM-1610 if necessary.) If your master is PCM-1610 format, then it is possible to trim the levels during the editing process by using the digital-domain level control located on the DAE-1100 editor console.

The tape format must be U-Matic professional NTSC standard. Best results seem to be obtained using a Sony BVU-800 or Sony V0-5850 VCRs, mostly because of their exceptional tape handling capabilities. Other ¾-inch U-Matic transports may be used, but care should be taken to make sure that the tape produced is without compromise. The BVU-800 is the machine used to reproduce the tape at the production facility, so special care needs to be taken to ensure that the audio heads reproducing the timecode are in the proper relationship to the video heads. I have seen machines that are as much as 13 frames off; in other words the timecode read by the audio head references a video frame 13 frames away from the one I am viewing. In reference to the Compact Disc, this means that you would not hear the beginning of a music when cueing up a specific selection on your CD player.

The blank tape must be a professional broadcast quality stock, with the lowest possible drop-out rate. Sony KCA-60BRD tapes have been through extra quality control steps to ensure the quality necessary for digital audio. Some record companies have been using the 3M Color Plus U-Matic tapes with excellent results. Whichever videotape you decide to use, test

it thoroughly before you ship a defective CD master tape half way around the world. This would be very embarrassing, not to mention the extra cost involved in doing it over again. Murphy says that it won't happen until you have a pressing overnight deadline to meet, but it's not worth taking the chance.

The biggest problem with tapes in the past has been "phantom drop-outs." The tape plays back perfectly during preview, but is plagued with errors when you try to cut the master disc. "But boss, it was okay a minute ago!" When you play the tape back again over the same spot, everything is perfect. This little inconvenience is usually caused by residual oxide or dust on the surface of the tape. The highest quality tapes and a clean tape machine will keep this type of error to a minimum. Remember, "Cleanliness is next to Digital Audio." No matter which machine/tape combination you use, make sure that the equipment is cleaned and aligned regularly, and that you know what you are doing. It is very easy to damage the delicate video heads during the cleaning process if it's not done properly. Keep the tape stock that you plan to use at the same environmental conditions as the tape machines. Temperature changes cause the tape stock to shrink or stretch during recording, while humidity changes cause condensation to form on the videotape, which leads to head clogging in the tape machine. Moving the tape into the room with the machine at least one hour prior to use will minimize these effects.

The PCM-1610 and the video transport must be turned on at least an hour before use so that all of the electronics are stabilized, especially the A/D and D/A converters. Some facilities have noticed that if the video transport has not been used for a few days, that a small amount of oxide builds up on the video heads, and the drop-out count is greater during the first half hour or so of operation. To be on the safe side, just pop a tape into the machine and press play during the warm-up period. The tape playing over the heads will polish off any oxide that had poor enough judgement to deposit itself while you weren't watching.

The next step is "black striping" and timecoding the blank videotape. There are two ways to perform this function. The first method is to record timecode and video signal from the Sony PCM-1610. With the unit set to the "dubbing" mode, it will produce a "digital mute" picture for the video transport. The second method is to record video black from a stable source, such as a video signal generator, with the timecode synchronized to this video signal. It doesn't matter what time the timecode generator starts with, as long as it doesn't roll over 00:00:00 during the tape. The timecode generator must be set for 60 Hz Non-drop frame timecode. Although this is done automatically when timecode is generated by the PCM-1610, it must be selected manually when using an external SMPTE code generator as in method #2.

Timecode must be recorded on audio track #2 of the U-Matic machine. The purpose for this step is to provide a stable, un-interrupted video sync pulse and SMPTE code from the beginning of the tape to the end. Why? Because the digital audio editing is performed as insert-type edits, which means that the video control track and timecode track is left undisturbed while the edits are made only on the digital audio (video) information. You should have quite a few of these tapes ready ahead of time so it doesn't take an extra hour every time you want to edit a tape for a CD master.

Do *not*, under any circumstances try to produce a black-striped tape by making a video copy of another black striped tape! I have seen the results, and it will *not* work. It would be easier to buy the right equipment using Xerox copies of hundred-dollar bills. (Hmmm?)

The tape containing the final mixes is placed in the playback machine and the black-striped tape placed in the record machine. The Sony DAE-1100 editor is set to perform insert edits on the video channel only. The PCM-1610 processor is set for digital signal input, and the sample rate switch to 44.1 kHz. There must be a *minimum* of 90 seconds of lead-in recorded before any program material may start; this 90 seconds must be PCM-audio with no modulation. With the PCM-1610 input switches selected to dubbing, and the playback deck halted, you will get exactly what you need at the record deck. Remember that 90 seconds is the minimum, although two minutes is even better and seems to be the nice, round number that most facilities have settled on. Write down the first valid timecode on the tape.

. . . continued on page 45 —

For additional information circle #32

THE DIGITAL FUTURE

— *continued from page 42 . . .*

You will need it later.

Digital Editing

Okay, you're really cooking now! You have 120 seconds of pure digital nothing. On to this you only need to edit the rest of your album. When recording a selection on to the destination machine, always record a little past the next edit point; i.e. where you want the next tune to begin. This is a good habit to get into, since it assures that there will be no little surprises when you play back the tape to verify what you have done. Without an overlap, you could end up with a void of a frame or more where there was *no* digital audio signal at all.

Note that after you finish adding each tune, it is a good idea to go back and listen to your edit. It is better that you check now, than to find out after the last edit that you botched the very first one. You usually can't fix one edit at the beginning without throwing off all of the others that follow. Write down the timecode at the start and end of each selection; these times will be required later. Also keep in mind that since there is no side one and side two, the extra space you left between the last song on side one and the first song on side two needs to be removed during the transfer/editing process. Finally, there is required a minimum of 30 seconds of silent PCM-audio lead-out after the last tune. Here again, I suggest 60 seconds or more. Write down the timecode at the end of the last selection, and the last valid timecode value present on the tape.

PQ Subcodes

The "P" and "Q" do not stand for anything; they are just letters assigned to the codes. There are, in fact, P, Q, R, S, T, U, V, W, X, and Y codes that are assigned to functions such as alphanumeric titles for display on future players, still-frame graphics for display on your television, and a plethora of yet undecided uses.

SMPTE timecode numbers you kept track of are necessary for the production of the PQ-Subcode information that will generate the P and Q channel data on the Compact Disc. The subcode information contains the total number of selections on the Compact Disc, the beginning and ending points of each selection, index points located within a selection, pre-emphasis on or off, and the end point of the entire program. The data necessary to produce this PQ-Subcode information is entered into a Cue Editor, such as the Sony DAQ-1000 or new Philips LHH0425, either at the mastering facility where the editing was performed or, more likely, at the Compact Disc production plant.

The information entered into the Cue Editor is recorded on audio track #1 of the U-Matic tape, starting 30 seconds prior to the beginning of the digital program material. The length of the PQ-Subcode data depends on the number of music selections, indexes, and other information required for your particular album. It may extend past the beginning of the recorded program material if necessary. The laser cutting equipment is smart enough to rewind the tape if this happens.

Because of the nature of the PQ-Subcode encoded on the audio track, it is also possible to generate the information with a personal computer. I won't even mention this fact, though. Why would anyone want to do this with a $1,000 computer when they can have a nice shiney $15,000 Cue Editor? Besides, I haven't quite finished the software yet.

The PQ-Subcode is read into the laser cutting machine just prior to the cutting process, and produces the Q data, or table of contents at the beginning of the Compact Disc when it is loaded into a domestic player. The table of contents contains the total number of selections, total time of the Compact Disc, start and stop times of each selection, location of any index points, two- or four-channel select, whether pre-emphasis is turned on or off, and the owner, country and serial number of each selection on the Compact Disc. The P data are generated during the cutting of the CD, and consists of the pause/start flags encoded at each of the specified index and selection start points during the program material. Got all of this? There will be a quiz.

If you do not have a DAQ-1000 or similar Cue Editor, then you will have to commit the information to paper. The CD manufacturer will usually supply you with the proper form required by his plant facility. The following data are needed:

1. Confirmation of the correct tape format, SMPTE timecode on track #2, and sampling frequency.

2. Program title, cassette and box identification numbers.

3.UPC (Universal Product Code), EAN (European Article Number), or similar additional facilities providing this service by the time you read this. By the way, this Sony/JVC transfer is done in the digital domain.

Transfers in the digital domain involving machines with different sampling frequencies at this time can only be accomplished using the Studer Sampling Frequency Convertor. The Studer SFC allows any format and sampling frequency digital input to be converted to any format and sampling frequency digital output.

If you plan to have Denon produce your Compact Discs, that facility transfers everything to computer hard disk (that's right, computer disks are spelled with a "k"), in order to intercept any "phantom drop-outs" that may occur. The company also is able, using this method, to produce a Compact Disc that contains more than 60 minutes of program material. Denon has its own method of entering PQ-Subcode information, so you must supply the SMPTE timecode information in written form. (I wonder if you could send your CD master on floppy disks? That's only 476.28 megabytes of data for 45 minutes of audio or, to better put it in perspective, 1,489 double-sided floppy disks from an IBM PC.)

The following short list shows the CD mastering facilities, the formats they except, and the PQ-Subcode they use to produce the P and Q channel information on the Compact Disc:

Manufacturer	Format	PQ-SUBCODE
CBS/Sony	PCM-1610	Sony PQ code
Sanyo	PCM-1610	Sony PQ code
Pioneer	PCM-1610	Sony PQ code
Toshiba/EMI	PCM-1610	Sony PQ code
Polygram	PCM-1610	Sony PQ code
Matsushita	PCM-1610	Own PQ code
Nippon Columbia	Own Format	Own PQ code
JVC	DAS-90	Own PQ code
Denon	Any Format	Own PQ code

Sony PCM-1600

The PCM-1600 is the precursor to the PCM-1610. If you use this processor you should be extra diligent in warm-up time and convertor alignments. Special care must be taken when you make a digital-to-digital copy of your master using this processor. If your master was recorded with "Emphasis On," but the emphasis switch on the processor is switched off during the transfer, you will end up with a Compact Disc that will have pre-emphasis on the recording, and will fail to notify the CD player to turn on the de-emphasis circuitry. Be careful, I think Murphy had a lot to do with this new technology.

Sony PCM-F1

Now on to something that everyone even remotely concerned about digital recording has been asking about. "Can I record or mix on my PCM-F1, and then transfer it to PCM-1610 for editing and Compact Disc mastering?" Well, the answer is a definite "Maybe." As with any choice of equipment for recording and mixing, there are both pros and cons to be considered before you make the big jump. The transfer can be made in the analog domain just as it can with any machine; transfers in the digital domain are a little bit trickier. The NTSC version of the Sony PCM-F1 uses a sampling frequency of 44.056 kHz, and a multiplexed sampling method where one convertor alternately samples one channel and then the other. The PCM-1610 samples both channels simultaneously at a rate of 44.1 kHz. At the time of this writing, the RTW Interface is the only digital interface between the PCM-F1 and the PCM-1610. I have reason to believe that a couple more digital interfaces and a PCM-F1 editing system will be shown at the Paris AES show in late March. [See Bob Katz' article describing applications of the F1 and F1/1610 transfers to be found elsewhere in this issue — *Ed.*]

The PCM-1610 will accept a digital transfer at the 44.056 kHz sample rate, but when it is played back at the 44.1 kHz rate it will replay faster and be higher in pitch by 0.1 percent. There is no sample-rate conversion performed, as some people have been led to believe. The difference in the sampling method produces a phases discrepency of approximately 80 degrees at 20 kHz, and five degrees at 1 kHz, as a result of the 13-microsecond delay between channels. This may or may not be a problem in your application, but you should be aware of the facts.

I guess that about sums it up from this end. Producing a Compact Disc is not a trivial matter, but when approached in a logical manner the results will be more than worth the effort. You should not be afraid to ask questions if you run in to any problems concerning your digital project. There are plenty of recording and mastering engineers who have already faced the same problems that you will undoubtedly encounter. You will not run in to one of them who will not take time out to help. Just try and not byte (pun intended) off more than you can chew by yourself. ■■■

Editor's Note: Readers seeking additional information on the various intricacies of digital recording and CD mastering might like to contact Roger Nichols directly c/o Soundworks A/V Studios, 254 West 54th St., New York, NY 10019. (212) 247-3690.

VISUAL MUSIC begins on page 53 —

news

AVC SYSTEMS MOVES TO NEW LOCATION

AVC has opened a new facility in Minneapolis. Formerly Sound 80 Recording Studios, the 12,000-square-foot facility has been arranged to provide demonstration capabilities approximating real-world conditions. Included will be a complete recording studio control room sound reinforcement system demo, and an AV/Teleconferencing demo room. Also, the Rodger Dodger Music Co., specializing in synthesizers and electronic drums will be occupying part of the new complex with their own separate sound-isolated demo studios.

The company's new address, which is in addition to its Chicago location, is 2709 East 25th Street, Minneapolis, MN 55406.

. . . **MORE NEWS** *on page 58* —

HARRISON FOUR-PAGE INSERT FOLLOWS ON PAGES 49 thru 52

INTRODUCING

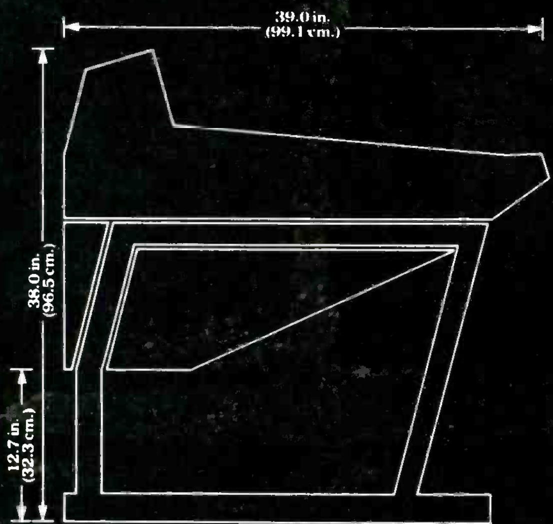
THE RAVEN

BY · HARRISON

THE RAVEN. A DREA

M BECOMES REALITY.

THE RAVEN. SPECIFICATIONS.



End View of Raven

Power Supply Dimensions

Dimension	*Inches*	*(cm)*
Width	19.0	48.3
Depth	17.5	44.5
Height	7.0	17.8

Line inputs:

Balanced	**Electronic active**
Nominal level	+4 dB, +6 dB, +8 dB (ref. 0.775 volts rms)
Maximum level	+30 dB (ref. 0.775 volts rms)
Actual load impedance	10 kohms
Expected source impedance	1000 ohms or less

Line outputs:

Balanced	**Electronic active**
Nominal level	+4 dB, +6 dB, +8 dB (ref. 0.775 volts rms)
Maximum level	+24 dB (ref. 0.775 volts rms)
Actual source impedance	40 ohms standard 20 ohms optional
Minimum load impedance	600 ohms standard 300 ohms optional

Meters:
Illuminated "pointer type" VU

Housing:
All aluminum alloy construction finished in multipart urethane finishing system, box steel welded legs, black "ebony type" wood trim.

Connectors:
Microphone input, line inputs & outputs—12 pin *Molex* connector module internal—*DIN* Eurocard 64 pin.

Power:
See reference manual for power and thermal considerations.

These specifications represent an overall system and not the individual components of a system. These specifications are subject to change at any time. *Important:* See reference manual for conditions of test and compliance.

Frequency Response:

Microphone input line output. Gain shall not deviate from gain at 1kHz for any frequency from 20 Hz to 20 kHz more than +.2 dB to -2dB.

Line input to line output. Gain shall not deviate from gain at 1kHz for any frequency from 20 Hz to 20 kHz by more than +.2 dB to -1.5 dB.

Noise:

Microphone in. The equivalent input noise shall not exceed -129 dB (ref. 0.775v).

Crosstalk:

Microphone in to line out. The crosstalk shall not be worse than -65 dB with reference to the driven output at any frequency up to 15 kHz.

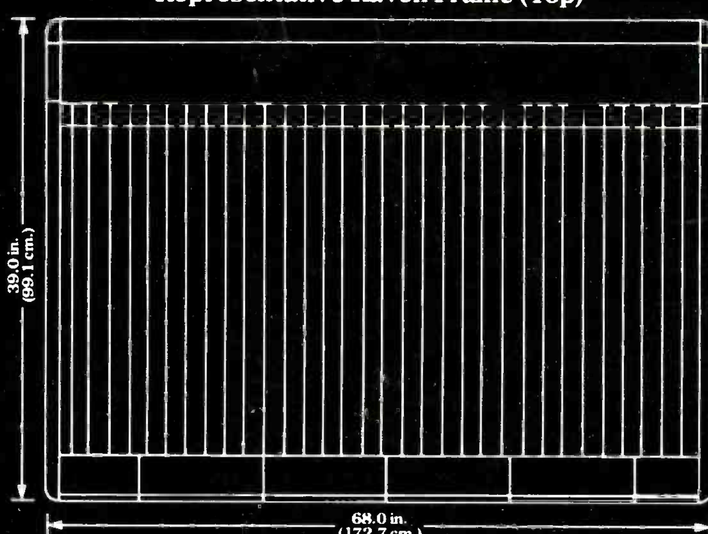
Distortion:

Microphone in to line out. The measured intermodulation distortion shall not exceed 0.1%.

Microphone input:

Balanced	**Electronic active**
Minimum gain (voltage gain) (fader unity, pad -20 dB)	Zero dB (unity)
Maximum gain (voltage gain) (fader unity, no pad)	70 dB
Nominal input level (fader unity, no pad)	-20 dB through -70 dB (ref. 0.775 volts rms)
Maximum input level (fader unity, pad -20 dB)	+30 dB (ref. 0.775 volts rms)
Expected source impedance	150 to 200 ohms
Actual loan impedance	1200 ohms

Representative Raven Frame (Top)



Harrison

HARRISON SYSTEMS, INCORPORATED
P.O. Box 22964, Nashville, Tennessee 37202
(615) 834-1184, Telex 555133

VISUAL MUSIC SCENE

The start of a new column that will examine the continuing impact of Music Videos, and the contributions that can be made by audio engineers.

This Issue: Post-production for Philip Bailey's "I Know"

by Adrian Zarin

At first glance, there wouldn't seem to be much to talk about on the subject of audio for rock video. After all, it's all taken from records, isn't it? The artists, producers, and engineers do their work in the recording studio; the finished product is then handed over to the video producer and director as a *fait accompli*. Their job is simply to fit visual images to the sounds they are given, right?

Well, it isn't quite *that* simple. If you've been checking out the latest rock-video and visual-music offerings, you will have noticed that the emphasis is more than ever on storytelling, as heavyweight film and TV directors are drawn into this burgeoning medium. They have brought with them not only the visual craft, but also the *audio* craft, of film and television production, including dialog and sound effects. From the flicking switchblades and snapping fingers of Michael Jackson's "Beat It" video, to the sounds of war in Paul McCartney's "Pipes of Peace" offering, hardly a new rockvid appears that doesn't have at least a few obligatory car sounds included on the audio track.

But these long time staples of audio-for-video and film post-production appear in a new light in rock video. They have to make their presence felt, but not compete with the music — which, after all, is still the main event. Doing this properly requires new skills, and a good understanding of both record engineering and audio engineering for the visual media. Which is where this new column comes in.

To kick things off, we'll turn the spot light on some recent clips that have made use of innovative audio techniques, focusing on one of the most ambitious to date in terms of non-musical audio elements. The video for Philip Bailey's song, "I Know," was created by director Jack Cole and producer Paul Flattery for Flattery/Halperin/Cole Productions. Although the piece employs a sound montage, effects and dialog, what really sets it apart from the rest is the fact that it's almost certainly the first rock video to superimpose dialog over the actual song, breaking what had formerly been the cardinal rule of music video: "Don't mess with the music."

Working with video editor Larry Bridges of Red Car Editing, and audio engineer Tom Davis, Cole and Flattery took Bailey's fairly innocuous love song and made it the basis for a full-blown mini-movie. According to Cole, the dialog was added to Bailey's music in order to help move the story line along. And while the idea to add it came up after production began, the dialog was nevertheless planned out with excruciating care.

"I planned and wrote the dialog to play into about a 17-second hole," Cole explains. "The section of the song I chose for the dialog was basically instrumental, with some repetitive lyrics that had already played out in the song. I thought it would be better to use this material as background, rather than a forefront piece this time around. The editor and I literally timed out the play of dialog so that it would fall between lyrical pushes. We handled it almost like a commercial that way. The dialog worked out so that it meshed nicely with the music, rather than just sitting on top of it."

For those of you who haven't seen the piece, a brief plot synopsis will help clarify some of the production details that follow. An opening audio/visual montage establishes the past of the character Bailey plays; he's a former high-school track star whose career has been aborted by an auto accident. Some brief dialog in a taxi depot identifies the character's present occupation: cab driver. *Then* the song begins. For about the first half, it's pretty much the standard rockvid fare as Bailey drives around L.A. lip-synching the lyrics.

The plot thickens when he picks up an obviously rich, important woman and her entourage. The woman, it turns out, is the main character's high-school girlfriend, Nikki. She has become a successful and busy actress, as is established by some dialog — spoken over the music — between Nikki and her agent. Nikki and Bailey recognize one another. Flashback to scenes of their romance. The cab arrives at its airport destination and Nikki is whisked away, with obvious reluctance, by her agent. It looks as though the piece will have an unhappy ending, but Nikki turns up at the taxi depot in the final scene. More dialog over the music between herself and Bailey's character. They still love one another. Happy ending.

Location Dialog Recording

According to Tom Davis, production dialog recording for the clip was handled in accordance with standard film procedure. "We did the cab depot scenes with a shotgun mike and fishpole boom," he recalls. "It was done just like film dialog; there was no music track or anything. It was a little tough because the place where we were taping was noisy. It was late at night, and you could hear a lot of crickets chirping through the open door of the depot. We sweetened the dialog quite a bit though,

THERE'S A REASON THE 360 DIGITAL KEYBOARD SOUNDS BEST.

Keyboards with real sounds have become one of the hottest topics today. Since 360 Systems introduced true studio quality reproduction, some companies have abandoned their products, while others have hired on Madison Avenue. There's even one keyboard that doesn't exist, even though its advertisements do. Another claims blinding intelligence, tomorrow's musical features today, electronic genius, and more! Like stratospheric price tags and Top Secret delivery dates.

In spite of this blizzard of misinformation, the 360 Systems Digital Keyboard remains the best choice for most musicians today. The reasons are all about music, quality ... and the price.

There's a huge quality advantage to getting 360's pre-recorded sounds. We sample a large number of notes across the keyboard, so there are none of those speeded-up "munchkin" sounds, and no sad low notes with slow attacks. And we're the only keyboard with perfectly adjusted pitch, loudness and brightness for every single key. The 360 has exactly the right timbre across the whole keyboard, and perfect uniformity from one end to the other.

360 offers a big library of musical sounds, all stored on plug-in memory chips. Whether you are scoring a feature film or playing in a rock & roll band, you can customize your keyboard for yourself. Get as few sounds as you want, or have as many as 32 on-line for instant use from the front panel buttons.

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and were able to mask the cricket noise."

Recording the dialog inside the taxicab was more involved, but still standard film procedure, as Davis explains: "The cab was mounted on a process trailer, which sits very low to the ground. You put the vehicle on the trailer and there's plenty of room for lights, cameras, everything. The whole thing is towed by a truck. As I recall, I set my Nagra recorder right on the hood of the cab. I ran my cables through the window and placed a Sennheiser MK416 on a little shortie desk stand right on the floor of the cab. I put a sandbag on it, and pointed the mike right up at the actress [playing Nikki's agent] who was reading the lines."

When it came to deciding on acceptable levels of background noise on the production dialog, however, the project started to deviate from standard film practice. The dialog Davis captured on location was fairly clean, but not clean enough for this purpose. "It probably could have been used if it wasn't a rock video," he says. "Except for the fact that we were going to have this really strong music track, and wanted the dialog to cut through without having to play with the music much. That's where the real difference lies between doing this and doing a film with underscoring. Underscoring is just that — it remains comfortably *under* the dialog in volume level.

Because of these special requirements, Davis took the precaution during shooting of bringing the actors into more favorable acoustic conditions (a nearby trailer, etc.) and making a tape of them reading their lines wild (i.e. not to picture). The relatively small amount of dialog in the piece, added to the fact that very few lines are coupled with full-face shots of the actors, made this a practical approach. In addition, when the actress who played Nikki's agent was called in during post-production to add some additional lines, Davis took the opportunity to loop all of her dialog.

Audio Post-Production and Processing

Post-production for the clip, handled at Pacific Video in Los Angeles, provided the opportunity for quite a bit of creativity in the area of non-musical audio. First of all, there was the opening montage to assemble. Director Jack Cole, having created montage sequences for feature films such as *Mahogany* and *Fantastic Voyage*, is something of a specialist in this area. Many of the decisions on the soundtrack for the opening were made in post-production as the montage took shape. A heartbeat, cheering crowds, screeching tires and other sounds mate with visual images to tell the story of the main character's aborted athletic career in well under a minute. Davis used stock sources for most of these effects, modifying them to suit the needs of the piece; for example, the tape speed on the crowd sounds was varied for a surrealistic effect.

Davis decided to build a composite stereo post-production mix rather than the separate music, dialog, and effects mixes common in film and TV work. "I felt we should approach the mix more like a record situation, as opposed to doing music, dialog, and effects tracks separately," he explains. "In this case everything was so dependent on everything else that I saw no reason to separate them. Besides, since we were dealing with stereo, I would have had to have six or seven different master tracks [stereo dialog, stereo effects, stereo music, etc]. I didn't want to get into the tape hiss buildup that would involve."

The music portion of the audio program was handled in the standard manner for rock videos. Bailey's record company furnished Davis with a 15 IPS, two-track stereo copy of the song with final mastering and EQ changes on it. He then transferred that tape onto a four-track machine with timecode, which would be used during final mixing and editing of the clip. He also made Nagra copies for use in the field (lip-sync scenes, etc), and a 16mm mag track for use in preliminary editing of the film stock on which the clip was initially shot.

Processing presented a special set of problems during post-production mixing. Besides having to give a unified sound to the panoply of production dialog, wild lines, looped dialog, and stock ambient sounds (traffic noise, room noise, etc), all of these sources also had to be processed in such a way as to be compatible with the music track.

"I believe we compressed the dialog quite a bit more than we would on a standard film or dramatic TV project," Davis says. "We equalized it a little bit tighter, too. When you listen to the dialog track by itself, it sounds very harsh, very unnatural. Again though, it was similar to mixing instrumental tracks for a record. A lot of times a given instrumental track, heard by itself, will sound *very* odd; but in the whole mix, it sounds great. That's the way this ended up. The dialog was very 'brittley,' very compressed — it really 'smacked.' But because of this, we didn't have to dip the music level much at all."

Given the nature of the project, dipping the music level was another touchy area. "Since this was so new," Davis says, "we were not real sure how we ought to do it. The question that was constantly before us was, 'Well, how much do you mess with the music track?' The biggest problem was making a smooth transition in dipping the music for the dialog, and then coming out again."

The production team experimented with a variety of approaches — including radical changes in the music volume — before devising a mix that they felt did justice to both dialog *and* music. "What we ended up doing," Davis continues, "was playing with the EQ and compression even more in order to make the dialog really cut through. As it turns out, we hardly touched the music levels at all. I kept the composite mix just a little bit down in volume throughout the whole piece — only a couple of dBs or so. In other words, when the track was really pumping, it wasn't peaking as far as one might normally peak it. This way, I had a little bit of headroom left, so that when something else came along, it would just kind of step over the top. I didn't have to pull anything back to accommodate."

Davis did find it necessary to push the music volume up slightly at the very end of the piece, after the last line of dialog is spoken. The final scene contains both production dialog and wild lines recorded immediately after the taping. To cover the transitions between these two sources, and mask the undesired background noise on the production dialog, Davis had to add some room ambience to the mix. "When you pulled all of that out after the last line of dialog," he explains, "It was necessary to 'goose' the music a little to compensate."

Multitrack Remix for Video

While the "I Know" project did not involve any remixing of the song itself, special "video mixes" of sounds would appear to also be on the cards for future rock videos. Viewers and technicians alike often complain of what Davis terms the "Tommy Effect" — named for Ken Russell's film of The Who's rock opera — in many of today's videos: an abrupt cut from an extreme close up of the lead singer lip syncing the song, to an extreme long shot of the same singer atop a distant mountain. The vocal, however, remains right out in front of the mix, and right on top of the listener. In other words, the sound lacks the perspective that the picture gives you.

An alternative to this is to remix the song especially for the video, much as songs are now remixed for dance clubs, radio, etc. "I would love to do final mixes of a song after the video is cut," Davis enthuses; "actually do a music mix to picture, and *then* do sweetening and whatever. That way you could give perspective to the mix based on what you're seeing. For example, if you're going to cut to the drummer playing a fill, why not push the drum fill up in the mix to give a musical perspective to the video? If you're going to go out and do a story piece, then the sky's the limit in terms of what you can do with the music."

In the case of at least one recent video, Michael Sembello's "Automatic Man," a special video mix was produced that actually grew out of a special dance mix of the original song. Sembello originally went to EFX Systems in Burbank — which specializes in sound effects for film and TV — to spice up the dance mix of his tune with some percussive explosion sounds and other effects. When it came time to do the video, which details the literal falling to pieces of the song's robotic protagonist, it seemed natural to return to EFX and take the idea even further.

Working with director Irv Goodnoff and producer Alexis Omeltchenko, George Johnsen of EFX compiled an effects track that contains over 500 explosions and electronic "zaps," and which draws on the full gamut of the effects specialist's art, from field recordings of actual explosions, to processed natural sounds and digital synthesis.

With various video and audio tape machines in use, synchronization for the project was rather extensive, as Johnsen details: "The original song was on two 24-track machines. There was an additional 24-track with the dance mix slave [containing] sound effects for the dance mix, and another 24-track which carried all the sounds for the video effects. In all, there was a total of four 24-tracks used to mix the video."

As this project makes clear, engineers involved with tomorrow's video mixes will need to be very well-versed in synchronization and computer-based editing techniques. Unfortunately, these are areas in which many record engineers — the people who, because of their strong audio background, are likely to be handling these special mixes — have had relatively little experience.

"Nobody *really* understands timecode and how it's supposed to work," Johnsen says. "It really hasn't been with us that long. Video editors understand it better than anybody. We people in audio need to catch up with them."

"The record industry has really just recently gotten very sophisticated in terms of microprocessors and computers," Tom Davis concurs. "It's still really common to do razor blade edits, and to roll things manually. Television has gotten more sophisticated because there's so much *more* to television audio. When you get into synchronizing many different machines, there's a lot of other parameters you have to be aware of. Record engineers need to become more attuned to things like timecode — when it's okay *not* to worry about having some kind of synchronous timecode base, and when you *have* to have one. It hasn't been necessary in the past, but now record engineers will have to get used to electronic editing, and just letting computers do things for them."

As proof of his point, Davis cites an incident that took place while he was working on Van Halen's latest video, "Jump." The clip was shot and edited before a final mix of the song was available, using a fairly well-developed rough mix. When it came time to sync up the final mix to the edited videotape, everything went smoothly . . . except for one thing. There was one shot that featured a close up of singer David Lee Roth emitting a heartfelt vocal scream. The only problem was that the scream had been *deleted* from the final mix.

The band were adamant about keeping the shot in the clip. The easiest remedy would have been to go back and retrieve the scream from the original multitrack master of the song; but that was unavailable at the time. Davis' only alternative was to edit the entire music passage containing the scream from the rough mix onto the final mix. Fortunately, the stereo perspectives and overall sound of both mixes were

RCA/COLUMBIA UNVEILS NEW MUSIC VIDEOCASSETTES

Included amongst the January releases from RCA/Columbia Pictures Home Video were two music-video cassettes. *Eurythmics — Sweet Dreams (The Video Album)* features David Stewart and Annie Lennox performing "Sweet Dreams (Are Made of This)" and "Love is a Stranger," plus 12 other songs. The video album, produced by Jon Roseman and directed by Derek Burbidge, is a combination of in-concert performances, conceptual video clips, and animation.

Daryl Hall and John Oates — Rock 'N Soul Live, directed by Marty Callner, was taped during the recent H_2O tour and includes "She's Gone," "Family Man," and "Maneater."

similar enough to make that possible.

Working once again at Pacific Video, Davis located points at which to punch in and punch out, and set up the edit. The final stereo mix of the song, which had just been synchronized to picture, was on two tracks of a 24-track Ampex 1200; the multitrack, according to Davis, was "completely tweaked to do a flawless punch in and punch out." The stereo mix, which had been used for shooting, and which contained the passage with the vocal scream, was layed on a four-track machine.

"I used a computer to synchronize the four-track machine with the 24-track machine," Davis explains. "Before making the actual edit, I used Pacific Video's CMX preview switcher, on which you can actually preview an audio edit of up to two tracks — you can switch, but not make the machines [physically perform the] edit. In other words, you can *hear* what the edit sounds like, but the machines will not go into record automatically.

"The edit sounded great, so I hit the red button and recorded the edit onto two open tracks of the Ampex 24-track. It was all synced to the picture because of the timecode. We then just took that whole mess and laid it back onto the audio channels of the videotape.

"Record engineers are always very surprised when I tell them I can drop in and out and make a 10-frame edit on a composite mix, and nobody will be able to hear it," Davis concludes. "But I did it on the Van Halen video. The machines have to be set up just right, and you have to have the right computer equipment, but it would have taken forever to razor blade something like that."

With regard to future developments in rock video, audio remixes and special edits for video, along with the addition of dialog and effects, may all be just the tip of the iceberg. Video directors Jack Cole and Paul Flattery have many ideas on where things may go.

"To speculate wildly about the future, I think we might get [rock videos] broken up into sections, where you can actually leave the music for a bit and then come back to the record," says Flattery. "The emphasis may move away from the soundtrack, by which I mean the [original] record. I'm not saying this is happening now, but it may in the future if rock videos become good consumer items as audio *and* visual products combined. Therefore you won't have a film of a particular piece of music — Philip Bailey's song, for example — it will become a thing in its own right by Flattery/Halperin/Cole *and* Philip Bailey."

"Seen from a long-range perspective, there are a lot of possibilities," adds Jack Cole. "For example, you can tie a number of songs from an album together in one long story line; there's no reason why you can't choose three or four songs and make a 15- to 30-minute piece. After all, if you look at a film like *American Grafitti*, it was just a series of vignettes tied together with a lot of nostalgic rock music. Story-line videos wouldn't be much different."

"Music videos are in a period of transition," Paul Flattery sums up. "They're moving from being simply promotional items, into being video music entertainment in their own right, I won't say that's happened yet, but it's certainly on the way."

Whatever specific forms emerge in the future of music video, it is clear that the audio portion of the program has already passed the days of "just play the record." The relative newness of and the need to respect the primacy of the music in adding other audio elements, will necessitate quite a bit of resourcefulness in both production and post-production. In future installments of this column we hope to keep the recording industry abreast of these new developments, and help bridge some of those gaps that now exist between record engineers and audio-for-video engineers. Stay tuned. ■■■

news

PRICE ANNOUNCED FOR dbx DIGITAL AUDIO PROCESSOR: BROADCASTS AT WGBH NOW UNDERWAY

The dbx Model 700 Digital Audio Processor is now being marketed at a suggested retail price of $4,600 — $400 less than the $5k originally estimated by the company, according to Lance Korthals, director of the companies Professional Products Division. An optional mike preamp module, which incorporates two preamps, is also available at a suggested retail price of $370.

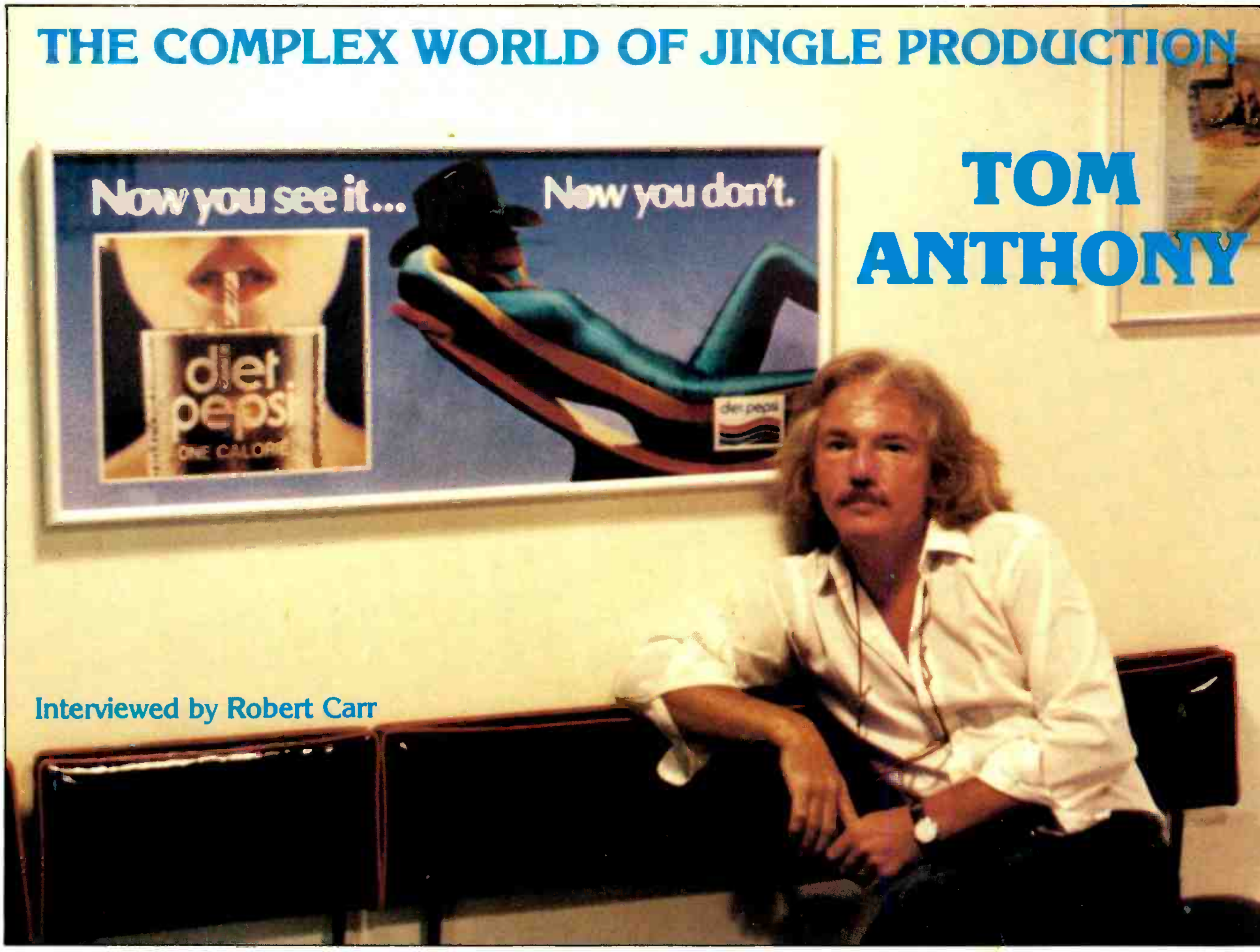
The company also expects the Model

. . . continued on page 199 —

THE COMPLEX WORLD OF JINGLE PRODUCTION

TOM ANTHONY

Interviewed by Robert Carr



Photography by Robert Carr

Unlike the specialized world of record producing, jingle production often requires that one man be responsible for writing, arranging, hiring musicians, producing, conducting, and even singing and playing an instrument on the date. In this realm, typical job titles defy the traditional boundaries. Such is the case with commercials producer Tom Anthony. Since childhood, Anthony has been pursuing his love for music in one capacity or another, through various styles ranging from classical to pop to jazz. He first earned a BS in music education as a student at SUNY Potsdam Music School. During the six years after graduation in the late Fifties, Anthony taught music in the public schools of New York and California, performed in a number of bands, and became director of the music educational services with the First Army Bands. Eventually he made the move back to New York, and went to work for Mitch Leigh's Music Makers, which at the time was the city's largest jingle house. Although it meant a cut in pay (to $125 a week), he accepted a job in sales and production (not writing) for four years to learn the business. Seeing that his future was "not written on Mitch Leigh's wall," he left with a fellow employee to form Ideas in Music. Unfortunately, the partnership was not as fruitful, and Anthony opened Tom Anthony Music in 1969.

The following interview took place in Tom Anthony's new location on Manhattan's West 46th Street, which opened in March 1983. The suite of offices and eight-track demo studio occupy the entire 14th floor of the building. His own private office/writing area was the subject of opening comments that served to break the ice, followed by some mention of the familiar jingles he's written for products such as Doublemint Gum, Miller Beer, Smith Corona, Burger King, AMC Jeep, Diet Pepsi, American Dairy, Coors Beer and scores more. The subject of matching the right music to a specific product provided the starting point for our conversation.

R-e/p (Robert Carr): I would assume that musical styles do not correlate particularly with specific types of products?

Tom Anthony: No. That gets into stereotyping the product, like "All dog foods are bouncy and cute." Stereotyping would be the death of the commercial. I would hate to think that all soda commercials had to be high-flying, with 80 singers screaming at the top of their lungs, and the strings and brass popping all the time. The audience needs a breath of fresh air once in a while. I refuse to stereotype products and stamp out the music in an assembly-line fashion. Jingles would all start sounding alike on the air.

The problem is that even though clients and ad people tell you they want something unique, they'll turn around and describe the same old thing. I have to fight that situation *all* the time. I could write a jingle like all the rest, take the money and run, but there's a certain amount of personal integrity that goes along with what I do. I have to use a mirror when I shave. I want to hear my songs on the radio, and go: "That's a great tune. I love that. I had fun writing it."

R-e/p (Robert Carr): So after you know what they want, you write the song to please yourself?

Tom Anthony: Obviously, nine times

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out of 10, they're going to tell me what they *think* they want, and I'm going to try to give them what *I* think they really need. That's my job. If I need an operation on my brain, I won't go to my dentist, or I won't try to do it myself, because a brain surgeon knows better. Likewise, it's up to me to take what the client and agency tells me, analyze it, and write the tune.

There's a critical analyzing process that I have to go through to write a jingle, and once I've analyzed the problem to see what kind of music is really needed for the commercial, I have to make it something unique so that people will take notice of it, remember it, and buy the product or service. I'm making the choice on millions of dollars worth of business, and usually, if I went along with what they thought was the right musical idea, I'd end up with another sound-alike. I would be just taking the money and running.

R-e/p (Robert Carr): I want to explore some of your thoughts regarding how to develop a musical concept for a specific product. Suppose I needed a jingle for my new product. What would you need from me to write an effective jingle for a national campaign?

Tom Anthony: I'd have to know what you're selling; who you're selling it to; what kind of a market it is — whether you're trying to appeal to red-headed girls from age 16½ to 18¼, or everybody in the world, because you don't care who buys it. I'd have to know how you want to sell the product, and what image you want to portray in the commercial, such as that of a "big corporation," or "young-and-hip," and so on. Very basically, you should give me an idea of what you want to say, because the lyrics have to be as condensed as possible. Then I take it from there, and try to write a jingle that's intelligent, attractive, entertaining, pleasing to the ear, memorable, and attention getting.

R-e/p: Let's consider an example: Suppose I needed to reach 16- to 19-year-old girls, as opposed to everybody. How would your approach to the two assignments differ?

TA: To enhance the appeal of a female-oriented product among older teenage girls, the first tendency would be to write a very up, bright, contemporary, funky kind of pop music piece. That's what I would do nine out of 10 times if it's a common, popular product like stockings, clothing, or something that doesn't need an image other than what you would normally associate with it. But there are times when you should stray from that sort of thinking, and do a completely different, left-handed curve that has nothing to do with the age bracket or the gender. Of course in that case, it helps to have an unusual product that can benefit from that approach.

When I wrote "Ooh, La-la, Sassoon," I thought the jeans would be bought by young, very fashion-conscious people. Up until that time, there were no jeans commercials except Levis, and that focused on a whole other area that had nothing to do with fashion. When I wrote the Sassoon music, disco was acceptable. I knew I wanted to center on that musical area, but still I wanted to come up with something a little different than the normal, everyday disco thing.

I knew the spot would be youth-oriented, but I also knew if I did it *right* — if I picked the right notes — it would affect everybody. What I wanted to accomplish with that ad was to write an incredible hook that could *not* be denied. Once it got in your head, you couldn't get it out with Raid!

What I ended up with was that five-note line. It was a very classy, sophisticated piece of disco music and, because of that, nobody had a bad thing to say about it. Three weeks later, every 85-year-old woman and two-year-old kid in the country was humming that sucker!

"I refuse to stereotype products and stamp out music in a production-line fashion."

They were just blindly soaking up the message and singing it to themselves in the shower, all over. The on-the-street recall was absolutely unbelievable.

R-e/p: When you're writing a particular song, do you keep in mind that the piece can be used in many diverse formats?

TA: Many commercials that have to do with a beer, soda pop, or anything in that category, are written with the idea that the song will be done a lot of different ways. Using the same music track with as many shots as these products make could not only prove to be very limiting, but also boring — there are exceptions. I never did *any* pool-outs [musical versions aimed at a specific market segment; for example, a country-western version, black version, Spanish/Chicano version, etc.] on my Diet Pepsi jingle ["Now you see it, now you don't"]. They all pretty much sounded like the original — because that had such a strong generic identity — that there was no reason to do the pool-outs; it sustained itself. And it was best to do it that way. People get to love those jingles. It's like, "Why are you messing with my favorite sound?" The audience becomes defensive... leave well enough alone.

For some reason, everybody thinks that if you're doing advertising for black appeal, for instance, you have to get really funky. For Spanish people you have to do a good Chicano piece of music. Not so. They all live in the United States. All black people do not tap dance; nor can all of them play basketball! As a community, by and large, they are basically the *same* musically as we are. You can't tell me that a black consumer can't listen to "Now you see it, now you don't," and not relate to it.

We definitely do those various versions on a lot of projects, but I can't say that there's always 100% impact.

R-e/p: You mentioned two very successful jingles — Sassoon and Diet Pepsi. With your track record, do you still have to go through the normal competition to get an account, or do the advertising agencies come directly to you?

TA: Very often I compete with other music companies. For a new campaign, agencies probably contact three to six leading music companies, and give each of us a certain amount of money to prepare a demo. The music company can do as little or as much as they want, from one piece of music to five, or anything from a piano/vocal demo to... I've done demos with 30 to 35 pieces. Very often I'll record a full-blown arrangement, because I know the people at that particular presentation think of demos that way.

R-e/p: So you have to be familiar with what the individual agencies expect when they say that they want to hear music demos?

TA: Right. Some people will say, "Just give me a demo that has you singing at the piano." But, depending on the type of project, that may not really be *enough* to convey the idea of the jingle. People think that piano/vocal recordings are clever and "artsy"; if a piece of music holds up at that level, it can certainly hold up as a full production. Well, they don't make allowances for rhythmic passages, or the groove. Very often, the rhythmic lick is what *really* makes someone attached to a piece of music, and you can't get that in a piano-and-voice demo. You can convey only the melody and chords, with very little attention to colorations or other really strong parts of the song.

I've found that a demo is much better if you take it at least to the level of a full rhythm section and a group singing so you get some of the power, the rhythmic feel, and some of the physical feeling of the music. I've had some demos as full as a small symphony, because the account executive wanted the client to hear these pieces in their actual air-quality form.

R-e/p: Does that added expense come out of your own pocket?

TA: No. The agencies pay for everything. All my costs are always covered; there's no speculation.

R-e/p: Where do you make the demo recording?

... continued overleaf —

TOM ANTHONY

TA: They're all done in the eight-track studio I have across the hall. The board is [Sound Workshop] 20-in and 8-out. I purposely built an eight-track studio with a small control room so I would never be tempted to install 24-track equipment, or larger. If I wanted a huge board, I'd buy one and run a studio. But that's a major rock tied to your tail. I *don't* want to be in the studio business!

The initial cost of putting in a major studio is something like $750,000. You have two maintenance engineers working eight hours a day, at least — there's $60,000 to $65,000 a year. You've got staff recording engineers, which is another big nut to meet. You have to update your equipment all the time. So the first two years you spend a million and a quarter, and you really have to keep that studio going 24 hours a day to pay for itself.

Too often people build studios for the *wrong* reasons: they build monuments, not facilities, and they serve egos more than any practical business purpose. All too often, all it does is give you one heck of a headache. It's like buying 10 farms, or 20 cars. What's it for?

The place I have is functional. I do all my demos here, and I have freelance engineers who work for me. That saves me from going out and renting studio time, which can eat up the *whole* budget for a demo. This way I can put a lot more into the demo itself. My eight-track studio has paid for itself on that basis alone. While music is an art, it is also a big business and has to be handled as such; you *have* to know what you're doing. I'm no different than anyone else. I want to make a few bucks, too. I just want to do it better than anyone else, and have more fun while I'm doing it.

R-e/p: Does that functional philosophy carry over to your choice of studios for completing the final version of a project?

TA: Just like every other business, you have a personal relationship with the people who run the place. You might have a financial arrangement based on the amount of work you put in there, because business sense is important, too. Suppose two studios are equal, maybe even identical. One owner says: "Okay, because you're going to be doing four or five sessions a week in here, I'll give you a rate of X amount of dollars for demos, and X amount of dollars for finals." The other owner may quote you a price that's $50 per hour higher. If you're going to be doing a couple of hundred dates a year, what's the sense of spending all the extra money?

But consider, too, the service organization — how they take care of you when you go in there, the follow-up service, how careful they are when they're making tape copies, and delivering goods to clients. You *have* to look at the whole picture, not just, "Gee, I love this board, because it can make a certain kind of sound." The equipment has to be good, don't get me wrong. And everything that you could possibly need has to be available to you. But whoever puts all those things together is the place that gets *my* business.

R-e/p: Where do you like to work in New York?

TA: I like to use a studio called Right Track on West 48th Street. I like the sound in the room; it's well built. One of their rooms has a magic. It's a good rhythm sound, and is nice and bright —perfect for brass and strings by themselves. Instruments come *alive* in that room. I like the equipment and the way it's maintained.

R-e/p: Since you seem to prefer a consis-

tency in terms of the studio you work in and the engineer you work with, does that consistency carry over to the other people you hire for dates, such as the musicians and singers? Or do you tend to cast them on a per session basis?

TA: I have my favorites, and I use them, because they're more exciting. One group of musicians may play note-for-note exactly what's on the page with no excitement. Another group can bring the same chart alive with their own personal ideas and input that they'll add to the piece without me having to ask for it. Sometimes I have to hold them back. Some players are better than others, and I like to put the best of them together.

R-e/p: Are there particular skills that you look for in the musicians you hire?

TA: Yes. And some things may not be as obvious as you want them to be, like being able to deal with the "time factor," as opposed to producing music for the record business, where people may sit in the studio for weeks on end completing one project. We accomplish the same thing in one hour, which is the basic unit of recording time in the ad business. At $70 per hour, I *don't* want people sitting around. We usually have about an hour to record the band, an hour for the singers, an hour to mix, and then it's good-bye. That doesn't mean it's "Slip-Shod Enterprises." It means you cut out the bullshit and learn to work fast. You don't get high or put blue bulbs in the ceiling and act funky. Yet you can still have fun, laugh, and enjoy yourself. There's just a lot of discipline in dealing with the time factor. That's where the producer's job gets delicate. He has to keep the recording session going at a comfortable pace, but also keep the excitement level quite high, so everybody's animated and happy.

R-e/p: I would think there'd be a fine line between keeping the players going, and getting them to be relaxed so the music feels laid-back, and in-the-pocket?

TA: Right. But these people are used to doing recordings all day, every day; there's not *too* much that shakes them up. They always have a good time and do a great job as long as they know that you, as the producer/writer, know what you're doing and what you want. If they think the person in charge hasn't prepared himself — has made awful charts and come without direction — they'll play terribly. Not on purpose, but you just won't get anything out of them; they're *not* going to make up a direction or essentially write the jingle for you. These are the very best musicians in the world, but they react to leadership like anyone else. The producer's degree of preparedness is reflected in the performance.

Also, your ease of operation, your standard operating procedure, is *critical.* I have a tremendous amount of fun in a session, but I know exactly what I want when I go in there — no questions. If I have to change the direction, or even rewrite the jingle on the spot, there's no problem, because I'll handle it right there. Whatever is necessary to get the job done will be done. That leads to happy, loose sessions that are a lot of fun, yet I get a tremendous product.

R-e/p: So basically you set up a strong form or mold for the way in which you want the session to proceed, and then let the players be free within those boundaries?

TA: That's right. Very highly educated adults — I don't care what kind of group it is — are *no* different than school kids; they need the discipline. You have to say, "This is the way we're going to behave in here for the next hour. As long as *you* know that, we can be happy. We have a lot of room to work within these boundaries, but this is the way it's going to be." That works wonderfully. Most people want the regimentation. Then they let the string out knowing what the boundaries are, and you get a much better performance.

R-e/p: Does that mean you write everything out note-for-note on the musicians' charts?

TA: For everyone but the rhythm sec-

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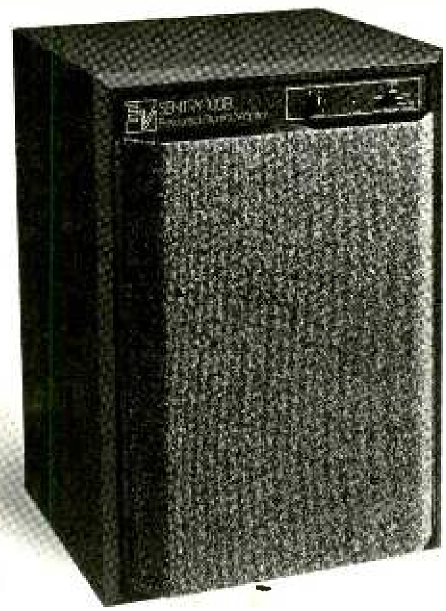

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SPECIALIST ENGINEERING SKILLS FOR COMMERCIALS AND JINGLE PRODUCTION

A Conversation with Jim McCurdy, regular engineer working with Tom Anthony

Jim McCurdy's first job after he quit high school was as a roadie for a small promotional band that played supermarkets around metropolitan Manhattan. Fortunately, he soon landed a promotion himself — to that of apprentice at a local production house doing radio syndication packages, which included ID jingles for radio stations, and custom advertising spots. Although he had the opportunity to test his skill doing some voice-over work and character voices, his attraction to the technical end of recording was stronger. A year or two later, he was engineering jingles and some record dates at O.D.O. — now known as Soundworks — where he stayed for a year, and then moved on to Six West Recording.

At Six West McCurdy first met and worked with Tom Anthony in the early Seventies. Another move to the Hit Factory three years later found him working with such artists as Dionne Warwick, Bonnie Raitt, and most notably Roberta Flack (*Feel Like Making Love* album) around 1975. McCurdy toured Japan and Australia with Flack doing live concert-sound mixing, and has since returned to Tokyo to engineer a live concert featuring Sadao Watanabe accompanied by the Tokyo Symphony Orchestra and a New York rhythm section.

In addition to his freelance record work at the Hit Factory and A+R Recording in the mid-Seventies, McCurdy served for a while on the staff at Sound Ideas, where he made some of the first digital recordings around 1976 and 1977. (Approximately 13 albums a year for two years on eight-track Denon digital machines.) In the last five or six years, he's completed roughly 40 jazz albums, mostly for release in the Japanese market.

McCurdy's freelance work has taken him to studios such as Sound Ideas, A+R, Automated Media, Sound Mixers, Sigma, Counterpoint, and Right Track. Most of his work currently focuses on jingles projects, with occasional record dates. He considers himself uniquely qualified to provide valuable insight into the differences between recording and production techniques for "conventional" record-company dates, and the specialist requirements of jingle and commercials production.

"Many engineers, producers, and musicians tend to look down on recording commercials," he offers. "I've done live recordings, live PA, studio record dates, commercials, film scores, and everything else. In my opinion, commercials are *the* most exacting and demanding type of audio engineering you can do. Not only do the final versions of jingles have to be excellent, but you have to turn out high-quality low-cost demos in a very short time. Demos are the first impression that a client gets about a particular musical concept, and a weak demo could easily lose a competition for a jingle project.

"In addition, clients and agency executives are purchasing higher-quality sound reproduction equipment on which to critique demos. The Pepsi people, for instance, have professional stereo gear, and *always* listen to presentations on big monitor speakers. For clients like that, I very often have to take a little extra time to prepare a more sophisticated, half-track, stereo tape that runs at 15 IPS, as opposed to a full-track, mono tape at 7½ IPS.

"Ironically, demos for commercials tend to cost a fortune, and in the past five or 10 years, budgets have gone down instead of up. In fact, the amount of money available has become a deciding factor in how effectively a music house can compete via demos for certain projects. When you're called to pitch for something like Pepsi, or any other national account, your demo has to achieve the *same* impact and listener reaction as the final version. To do that, Tom [Anthony] and I built a small demo studio at his office, and we can get some amazing results, even on a moderate budget."

Building an In-House Demo Studio

"The major consideration for us was money, which all came out of Tom's pocket. We had to look at each piece of equipment as though we had never bought equipment before — sort of looking at what we wanted with a beginner's mind. We found that the best approach to building a demo studio is to analyze every part of the whole operation and determine the needs versus the individual costs — right down to the type of mike cable and mike stands.

"We decided we wanted very good mikes, but chose not to buy new ones, because the prices were too high. Instead, we went with a full complement of used, top-quality,

TOM ANTHONY

tion, we'll often indicate a melody and chord progression on one staff, and then on a second staff below that we'll write a rhythm chart so the bass player and drummer know what's going on within any particular measure. If we just want straight time, we'll write hash marks and chords, and they play whatever they want within the feel of the tune. You don't necessarily have to write out all the notes, just the important accents and rhythm figures so the band can play together. With rhythm players, writing too much can sometimes work against you.

Of course, there are many times when only exact rotation can work — instances when everyone is playing the same line, or a classical piece, etc., etc. Obviously, then, its note-for-note.

But in a piece of music that's supposed to be funky, how do you tell a bass player what to play for an entire piece? I know a lot of people who try, but how can you think like a guitar player, a bass player, a drummer, or piano player? They're specialists on their instruments. My piano player is so brilliant, I could never come close to imagining a part that would be better than what he would play. But I still do an awful lot of telling him what to play. "Don't do that." "Keep your hand away from that section of the keyboard." "Bring the part down in the mid-range." "Just play rhythm."

R-e/p: Singers are probably the most versatile musicians you come across in the studio, but they have to blend closely with each other. How specific do you need to be with them?

TA: I'll often write out the singers' parts for them. But, just as often, I won't bother. Sometimes I can save time by telling them what I want, and letting them write it down for themselves. When the parts are obvious, I can just look at them and they'll sing it. Basically, it depends on how intricate the piece is. If it's simple, why bother going to all the trouble? A very involved choral part with a lot of four-part harmony needs to be written out.

R-e/p: Let's say a musician — a guitar player, for example — is not cutting his part. Would you get him out of the studio or let it go and fix the part later? What course of action would cause the least damage to the session, which obviously is structured to run as fast as possible?

TA: If it became obvious to me that he wasn't going to cut it, I probably would very politely excuse him to the side for a second, and let the other guitarist handle it. Or you can let him play the part, but just shut his channel down, and overdub it later. That's one of the aspects you have to deal with in commercials as opposed to other forms of


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condenser mikes for about half the overall cost of new microphones. On the other hand, the new condensed-format recording systems — eight tracks on half-inch — have gotten good reports and are relatively inexpensive. [McCurdy discovered that an eight-track, half-inch machine is almost 1/6th to 1/8th the price of a used two-inch, 24-track machine.]

"But because some of the spec sheets can be misleading, I talked to many people who used such [half-inch] tape recorders, consulted an old friend at a professional-audio retail shop about comparative qualities, and tested some machines myself. Once we got a good idea about what was available, we decided to go with Otari MX-5050s. Following that kind of economy thinking each step of the way, we can get great results and still keep our costs down."

Recording Demos for Commercials

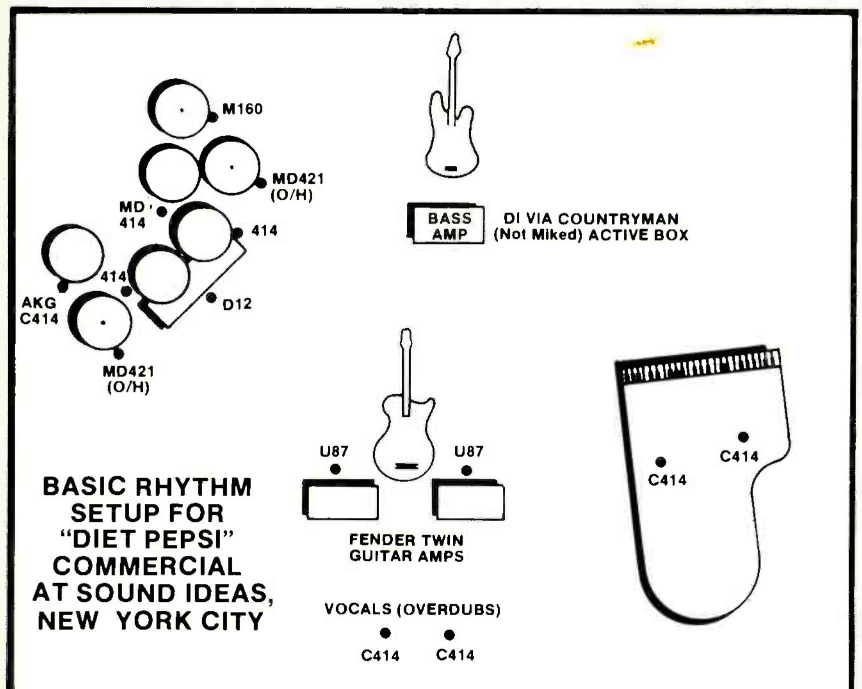
"Generally, the same miking techniques, instrumentation, and even the same players are employed on both the demo and the final versions. But doing a demo on an eight-track, of course, calls for some compromises. Usually, I record the band [drums, bass, guitar, and keyboards] in stereo on two tracks. That leaves six tracks for the singers. Because group singing is the most common, we may record three passes [one on each of three tracks] of the melody. Then lay down two passes of the entire group singing a harmony line. The last track [of the eight-track tape] might get a third harmony part sung once. For the final demo mix, we'd collapse all eight tracks down to a two-track machine and make our cassette copies from there.

"In jingles, the vocals are most important, and they get changed frequently. We might do a demo for a client who likes the overall concept, but feels there's a problem with some of the lyrics. That might mean rewriting the lyrics and re-recording them. If the vocals were all mixed together with the music on the eight-track, we'd have to re-record the music, too. This way, we just go back into the studio and punch in whatever line has to be changed.

"If the same vocal tag is used for all the spots in a campaign, it can be recorded once, and 'flown off' the multitrack to the other versions, so the singers don't have to sing on each one. I do it manually by trial and error. I advance the tape to the vocal beginning, back up the tape a few inches, and hit the start button on a convenient beat. I just count out the appropriate number of beats from that point, hit the play button on [count] four, and see how close to the start of the next measure the vocal comes in. When I know that, I just advance or retard the start of the tape accordingly. After you do it a while, you get a feeling for how much pre-roll you need."

Recording the Final Mix

"When we go to 24 tracks for the final, I distribute the band across the extra tracks. I assign the drums to four tracks [kick and snare on separate tracks, and two tracks for a stereo mix holding the rest of the kit, minus the kick and snare]. That lets me spread out the toms for a stereo mix if I need it, or easily collapse them for mono. Electric guitar and bass are recorded on one track apiece. I may split the acoustic piano for separate high and low



BASIC RHYTHM SETUP FOR "DIET PEPSI" COMMERCIAL AT SOUND IDEAS, NEW YORK CITY

TOM ANTHONY

recording that might be less demanding time-wise. But generally, the jingle players are highly qualified technically — they're killer sight-readers, and very flexible in terms of handling different kinds of music. They have to sing and play everything from full-blown rockers to very sophisticated, laid-back pieces. They can go from A to Z in tempo, volume, feeling, etc.

R-e/p: As the producer, do you prefer to stay in the control room, or go into the main room with the players.

TA: I hear better sitting in the control room, rather than standing out in the studio listening through a set of cans [headphones]. What the heck am I going to do out there, anyway? Conduct the rhythm section? [Laughs] It's like conducting the clock on your night table. What am I going to tell it? Click every second? Anybody can holler "1, 2, 3, 4," and count off the music. This is a *sound* business. I prefer to hear the music well.

If it's an orchestral situation, where some conducting and direction is really needed, then I'll go into the studio. But I'll stay there only as long as I really have to, and then I'm back in the control room. Again, I prefer not to have someone else conduct for me, because just like with my arrangements, I prefer my own conducting, too. It allows me to add my own emotional and physical input that only *I* know. How do you explain those intangibles to anyone else? As big as I get behind that baton, that's how big the orchestra gets. As emotional as I get, that's how emotional they get. Watch any conductor of any good orchestra. As he conducts, so the people will play.

Conducting is a leadership job, like a choreographer. If you stand out there in the studio like a rutabaga, and just tick off time with your hand in about three inches of air space, you're not going to get *any* excitement out of anybody. The music is going to lay there. Have you ever watched Leonard Bernstein conduct? He jumps around like a Goddamn gazelle. He's all over the podium. That's pretty much the way I do it, too. If you want somebody to get excited, you have to "goose" them a little bit. If you're not there with your thumb, they're *not* going to move.

Again, musicians really *want* direction and leadership; once they know the rules they can do anything. They also want leadership in terms of emotional content, physical content, and interpretation. Thirty-five musicians in a playing situation doing a particular piece of music with very specific emotions will *not* give you 35 simultaneous and identical interpretations. No way.

A lot of times you don't have to say very much. You can conduct it, and people will understand visually just how you want that piece to be played,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because you're literally drawing pictures in the air with your hands, your face, and the way your body moves. That's why some conductors stink, and others are great.

R-e/p: For inspiration, do you ever show a video of the commercial to the musicians and vocalists either before or while the music is being cut?

TA: No. I can explain what kind of spot it is. I'll ask for the music to be played a certain way, and explain how I want the music to feel. Then I'll possibly say, "What's happening on camera is a grown man riding a tricycle down the street with a bus chasing him, so play this way, or do this." That way everybody has a picture of what's going on, and they play accordingly. It's like being a director on a film.

R-e/p: How much guidance does the recording engineer need from you?

TA: I generally leave him alone, because I have a pretty good rapport with the engineers I work with. I've worked with Jim McCurdy for 16 years, and other engineers, like Mike DeLugg, are such talents that they understand what I want. The only time I need to tell them which way to go is when I want something specific that nobody else would possibly think of. [The article "Creative Art of Jingle and Commercials Production," published in the December 1983 issue contained many interesting comments from Mike DeLugg, who currently works as an independent at Lucas/McFaul's in-house 24-track Wharehouse Studio — *Ed.*]

For example, I don't sit on top of the guy when he's mixing. I take a walk around the block; go get a beer or something. I don't let him do the mix, but I at least let him get it going based on the information I've given him. What the heck am *I* going to do when he's setting up the board, and bringing it down to mixing status? That's a 20- or 30-minute process that nobody can help anybody with. Those engineers are certainly capable.

R-e/p: With regard to how to get the necessary sounds, do you leave that up to engineer Jim McCurdy, too?

TA: No. I'm very specific about how to get sounds, and how they're made. It's not really up to the engineer; it's the responsibility of the guy sitting next to him — the producer. Engineers know how to get all these sounds, but if you don't decribe exactly *what* you want, you're just not going to get it.

R-e/p: So you might give a direction such as, "I want the bass drum to sound like the kick on the new Chicago single?"

TA: You can do that, yet you really have to be more specific than that. Acoustic and natural instrument sounds are very easy to identify and figure out. But any new sounds that are out right now, of course, are going to be all electronic. They require a little bit of study to determine how they're arrived at. You really have to know the *basics* of the sound you want, and your ear has to be tuned to those sounds. Some of the nuances involved in the refining of that sound have to be sought after, analyzed and figured out, but it's not that difficult. You identify the basic waveform —sine wave, sawtooth wave, whatever —and then work from there. We spend a lot of time on synthesizers, and do a lot of experimentation.

R-e/p: Do you ever use a synthesizer to replace a string part?

TA: Not to replace strings; I don't like that. There's nothing in the world that sounds like real strings. If you specifically *want* a synthesized sound, you should use a synthesizer, but not as a replacement for or to augment a string section. And synthesizers really *don't* save you any money on a session.

Say you had four string players, and you wanted to augment that section with a synth part. For all the gear that synth players haul around with them, it costs $135 just for the cartage alone! Then you have to pay him double scale, because he's the only guy working, so he's getting $140 per hour. If he works only one hour, that's $275. You can hire four string players for that, and then you'll have eight violin players. God help you if the synth player works two hours!

I use a synthesizer for a lot of parts, because you can get unique sounds that are unavailable anywhere else. It works great as part of the coloration of another section — to beef up the rythm section, fatten up the brass, change the quality a little, make it a little phasey . . . whatever.

R-e/p: Are all the real string and horn parts overdubbed?

TA: A lot of times they are, but certain times I like to record things live. Very often with classical music you don't have a rhythm section to lay down first. Or with pop music, how are you going to change tempo — a little retard — and then catch it in the overdubs? There's also something to be said for that big, open-hall sound with the mikes way up

COMMERCIALS PRODUCTION — continued . . .

parts, and go for stereo with the Fender Rhodes if there's any kind of stereo effect available. The entire string section is laid down in stereo and kept that way in the mix whenever possible. Stereo horns probably would be more desirable in a classical situation rather than for a pop-music date.

"The brass, comprising sections of trumpets, french horns, 'bones and saxes, typically would be split up by section, and assigned to individual tracks so I can make little changes later on. For example: once the vocals are recorded, we might find that we want more trumpets, because they have the melody. Or very often the bass 'bone is kept separate. It generally plays a different part, and we might want to emphasize that in the final mixdown. If it's on a separate track I can bring it up.

"An engineer should use discretion when mixing in stereo for a jingle. The majority of the time buy [the amount of air time purchased for advertising] is usually on AM and mono FM stations, so I would normally make a mono mix for those uses. But the agency may buy stereo time on a few big FM stations, like WNEW or WPLJ in New York.

"I would want to make the stereo mix as exciting as I can, and utilize the apparent effects of center image and panning that are available when there are two channels to mix on. On the other hand, I don't want to pan things too hard left and right, because that same stereo mix will also be played back on some little FM portable radios that have only one speaker. I want the stereo mix to be semicompatible with a mono playback system, so I wouldn't go too crazy with panning."

Use of Special Effects

"The use of special effects depends on the nature of the spot. if we're doing an underscore for a 'high-tech-looking' spot advertising a new electronic watch, we'd spend the time to achieve whatever effects are necessary. In the case of more straight ahead rock and roll on the 'New Wave' side, I would take some time in the mixdown to set up the equipment to make the spot sound New Wave — maybe set up an echo chamber on a real short decay, and pre-delay the send; patch in a DDL [digital delay line] to use on some of the instruments or some of the vocals; get a slap-tape repeat for the lead vocal; or put noise gates on the bass drum and snare . . . things like that.

"Whatever I do during a mixdown or the actual recording is dictated by the style of the music, and the necessary time is usually allocated when we're planning the session. If it

"Anything that you're going to bother keeping on tape should be documented so you know what was in that take; you may end up using it."

high in the room, or the type of performance you get from a whole group in concert. You don't get those things in an overdub situation.

The jingle I just did for Kodak Film had about 30 pieces in the studio at the same time. I preferred to do that live, because it was kind of Baroque-style and pop combined that really had to be performed at one time. Plus, I had only so much time available to me in the studio. There was no overdubbing in that situation, but other times we layer the tracks. We do the rhythm section first, then the strings and the brass, then this instrument and that. We always overdub synthesizers.

R-e/p: I assume that you probably follow the same basic recording procedures on the final sessions as you do on the demo that preceded it. But sometimes demos contain small, often subtle, items and qualities that catch the attention of the client or agency, and must be included in the final. Do you have to keep very accurate notes about each demo recording for future reference?

TA: Jim [McCurdy, session engineer] or usually the assistant engineer makes extensive notes on the take sheet about all the elements that go into a demo, so that when we do the final we don't leave out something important. We may have to indicate changes in tempo or feel. I'll make some comments like, "Hold this take, because there are great guitar licks." Or, "The take was really exciting but we probably need to edit the last note, because it was played wrong." Maybe one take was a little more subdued than another take, or that one was a lot more animated than another, etc.

The client may come back when we do the final and say, "Look, I don't want this thing as animated as the demo. We have to lay it back a little bit." Or soften it, sweeten it, make it prettier, whatever. "It's like the take we did two takes before the one that we chose." Then you go, "Which one was that? What did we do"

Anything that you're going to bother keeping on a tape — that you're not going to erase or record over — should be documented so you know what was in that take; you may end up using that take. We have to have very complete qualitative notes as well as quantitative notes about the instumentation and so on. If we don't do that, we waste too much time going through all the takes trying to figure out which one we liked for a particular reason!

I just went through a situation that's a good example of what I'm talking about. I did a campaign for a client where the demo was really a very forceful, rhythmic track. For some reason when we got the final music married together with the film, it wasn't as exciting as the demo, but it went on the air anyway. Some pieces of music shouldn't be played twice. Sometimes they're better, more exciting, more spontaneous right out of the chute the first time. Two weeks later you try to duplicate the same thing and it ends up stiff, or not quite the same.

R-e/p: Do you sometimes use demos for the final tracks?

TA: Sometimes. I do *very* strong, air-quality demos. We just transfer the 8-track demos from my studio across to the 24-track for the final, but that doesn't happen too often. Only once in a while do we get a demo that is just so *powerful* that it shouldn't be re-recorded.

R-e/p: Since so much of the jingle business is tied to what's happening on the contemporary charts, do you find it necessary to study the styles that are popular?

TA: Definitely. I keep abreast of everything that's happening in music. I *have* to. I listen to a lot of records, but I don't buy them. I could empty my pockets trying to keep up with what's new. [Laughs] I keep the radio going, but not all the time. When I've been in the studio all day, the *last* thing I want to do is go home and listen to music. I'm around music and music people all the time, so staying current is a very easy process; it's a part of life. I *don't* make a project out of it.

R-e/p: Do you feel that your jingles are tied to the music that's currently popular?

TA: Not necessarily. The Diet Pepsi spot wasn't tied to the current chart. Most of the time, advertising is a very trendy business, because a good number of creative ad people and copywriters are copycats. They go to a movie, like *Flashdance*, and every commercial till the end of the year looks like *Flashdance*. In the past several months I've had no less than 12 requests for a jingle that sounds like a particular record by a famous Top-Ten group.

One agency, who shall remain nameless, took the song word-for-word, note-for-note, and substituted the name of a very well-known product for two other words in the lyrics. They wanted me to record it and put it on the air. I told them

COMMERCIALS PRODUCTION — continued . . .

isn't, then I do the best I can to whip together the project in the time that's alloted to me."

Timecodes, Clicks and Sync Pulses

"Doing an audio/video lock up with SMPTE is not the most common procedure. In the case of a television spot, a click-track referenced to the frame count of the film is used to keep the band in sync. As long as the musicians play correctly with the click, their parts stay in the right places relative to the action in the visuals. For syncing with film, we rely on a 60 Hz sync pulse, usually assigned to an edge track. If we do a full-coat [35mm mag], or two-track with sync so they can do a [sync-track] stripe, that sync is transferred directly to the four-track and the two-track master tapes. Isolation between tracks of the 24-track is very good. Crosstalk isn't really a problem as long as we print the click [on track #1 or #23] and sync [on #23 or #24] at around -10 VU."

Mixing Techniques

"I don't use automation. I find that it's usually faster to mix manually. Basically, Tom may make comments like, 'I'd like to hear more of the piano figure than guitars,' or 'The strings didn't really play as aggressively as I'd like. Can we do anything about that?' The music engineer should be able to hear what's *supposed* to happen, so the producer doesn't have to do a lot of directing. Not only does the directing waste time, but it breaks up the producer's concentration that may be used more effectively to guide the production of the overall project."

Selection of Appropriate Studios

"Tom has a list of about three or four places where he can work comfortably, and 'comfortably' is an *important* word. As in any other business, there are studios that specialize in a certain kind of service. The rooms that we use are targeted to do mostly advertising and jingle work. They may have a little area off to the side, but still in the control room, where a couple of the agency people can make 'phone calls during a session, and not be blown away by the audio level. Or little conveniences like having a charge account with a deli that can bring in food quickly for clients. Those little considerations make the job easier." □□□

No! Because the publishing company wanted a million dollars for rights, the agency wanted me to change it *just* enough so they could legally copy it. Well, there's a section in the copyright law that states you can't change four notes of a song and get away free! The agency just turned down the publisher for a million dollars. Now they're trying to record something that sounds *exactly* like what they tried to buy. We'd all be in court in no time! I'd be in Sing-Sing for the rest of my life.

But getting back to your original question, Diet Pepsi wasn't tied to the contemporary charts, and that was extremely successful. Therein lies the basis for one of my biggest arguments about "fad followers." That Diet-Pepsi piece was written for the campaign and for the pictures, *not* for any contemporary or sound-a-like value it might have. The music was strictly functional.

R-e/p: Did they bring you in to record the music and lyrics after the visuals had been shot and edited?

TA: I was in *before* they shot the visuals. They showed me exactly what it was going to look like, and that's why I did a more laid-back piece of music. I knew the pictures were going to be very hot, very sexy. If I had written a real funky, sexy piece of music, the networks would have kept it off the air for being *too* sexual. As it was, the agency had to put some of the girls in shorts instead of bathing suits.

R-e/p: How do you compose the music to begin with? Do you work with a Movieola or videotape, and measure it all out?

TA: I seldom look at anything more than two or three times. Generally, I don't even look at the film, because it hasn't been made when I write the music and record the demo. After the film is shot, I'll time the music exactly to the visual action, and record it in its completed form.

R-e/p: With the traditional 60-second television spots being replaced by shorter, 30-second versions, how do you manage to get all the information into such a small time frame, and still maintain the commercial's impact?

TA: You have to increase the amount of discipline in your writing. In a 60, you have time to write a bridge, and make it a mini song. In a 30, you can't just say the message faster — you *have* to have the same feeling of continuity. The whole piece has to tie together with a beginning, an ending, a build, and a story besides.

Within five minutes I can reduce a 60-second lyric to 30 seconds, and after you hear it you'll walk away with just as much information in your head as you did when you heard the 60 — maybe even more. Every component is so carefully designed that you cannot waste a syllable. You're allowed one or two adjectives here and there, but you *have* to get to the point. You have to learn to budget your language, and really hone everything down to the core of information that you're trying to convey.

The other half of the discipline is a matter of making the jingle attractive. The commercial can't be a piece of information set to music. It still has to be a fluid, inviting piece of information that really catches your attention. And

TOM ANTHONY'S PERSONAL-USE EIGHT-TRACK DEMO STUDIO EQUIPMENT LIST

- Sound Workshop Series 20 20-in/8-out console.
- Pair of Otari MX-5050 eight-track on half-inch machines.
- Auratone and CSI monitor speakers.
- Technics Model 1500 two-track mastering deck, and Mitsubishi DT-30 cassette.
- Tascam PE-40 graphic equalizer.
- Crown D-75 and Ashly FET-200 stereo power amplifiers.

TOM ANTHONY

the shorter the time frame is, the harder it is to do that.

R-e/p: Is there a songform that you've developed that works better within the 30-second format?

TA: You can still write two very short verses, each about two to four lines long. There's one sentence for the build, and a short hook. Then the guy talks for a while, the hook repeats, and that's it. That A-A-B-C form is usually about all you get out of it. Sometimes you find a strong enough little tune that pops along and doesn't have to repeat itself like a regular song. It doesn't build at all; it just lays there and does its job. Diet Pepsi is laid-back like that.

My Florida commercial follows the A-A-B-C form:

First Verse: "When the last leaf of autumn has fallen to the ground,

"And the icy winds through the empty trees make a howling sound.

"First it drives you indoors, and then it drives you mad,

"That's when you know you need it, and you know you need it bad."

Second Verse: "You say it every summer, 'What makes it go so fast?'

"And later on you wonder what makes the winter last.

"While the springtime you've been waiting for never comes somehow,

"That's when you know you need it, and you know you need it now."

Build: "You need the sunshine, and sea breeze

"Soft sand, and palm trees of Florida . . . Florida"

Hook: "When you need it bad, we've got it good.

"When you need it bad come to Florida.

"When you need it bad, we've got it good"

. . . fade. There's two verses, the build, and the hook.

R-e/p: As the list of adjectives gets cut down, does the music have to take their place — bear more of the responsibility of delivering the feeling and description?

TA: In a way I guess the music sort of replaces the adjectives with colorations. We have to be very strong and very pointed in image in terms of the musical mood and energy level we set. That means getting right to the point.

One thing we don't spend a lot of time on is the introduction: we set up an image immediately so the listener knows within the first two seconds *exactly* what he's getting into. Establish your energy level right away, like the drum fill at the beginning of the Doublemint Gum commercial, and make sure your opening lyrics are pretty strong. Right out of the gate you have to capture the people's attention, and sweep them along.

But getting back to your question, the lyrics pretty much tell you what musical area you're in, and the mood. Of course, you can control that to varying degrees. Instrumentation is really another word for "coloration," or "texture." It's like saying, "I want this painting to be red and blue," or "red and yellow." If I want the music to be liquid and flowing, I might use more woodwinds and strings. For the music to kick extra hard, I might use brass. Or strings can kick nearly as hard as brass if you want them to. Those are all little variations you have to ask yourself. It's really hard to say beyond that what color I might choose over another, because what I do depends so much on the style of the music.

"A commercial can't be a piece of information set to music. It still has to be a fluid, inviting piece of information that really catches your attention."

R-e/p: Do the ad agencies generally supply you with the lyrics, or do you write the entire package?

TA: The agencies supply me with the lyrics sometimes; I don't *always* like them. With a few exceptions, agency copywriters are not lyricists — writing is a whole other craft. Some of them go so far as to not even have a concept of time, and the number of beats between lines. They'll write a lyric and then read through it as though there was no punctuation — never leaving room for "2, 3, 4 . . ."

When you say words, or read them from paper, they sound a *lot* different than when they're sung. The meanings change; words that look perfectly sensible on paper can become corny or out of place in a song. I prefer to write my own lyrics, and I do on practically 90% of the jingles I write. I either rewrite what I get, or get a list of copy ideas and take it from there.

R-e/p: Other than a salesman, you are basically the jingle company. Do you think you would lose continuity in your product if you delegated jobs or authority to others to increase your output?

TA: The answer is yes, but with one exception. I do have an assistant who works with me side by side, day and night. His name is Gavin Spencer and he's a very talented, knowledgeable, complete musician. He came to me two years ago ostensibly to apprentice and, although he may have picked up a lot of what I do, I must say he brought a few things to the party himself. And who the heck wants to work alone anyway? I bounce ideas and get good input; he tells me when my ideas suck . . . "Hey, what about this instead," etc. He writes, sings, plays, understands, and listens. Finding a guy like that is rare indeed.

I'm sorry . . . *two* exceptions. Once in a while, I will get *so* busy, I just can't handle it all and still be effective, fair or healthy. Only then will I call in an arranger I trust implicitly, whom I know will understand and give me exactly what I want, almost as if I had done it all myself. Someone of the caliber of a Joe Beck or Alan Foust. My list is very short.

Having one guy writing lyrics, another writing music, someone else doing arrangements, and another guy producing is like passing a story through 10 people arranged in a circle. By the time the story gets back to where it started, it's completely different. People come here for me to do their jingles. If I hired people, they wouldn't be getting *me.* A lot of people do that, but I don't like it.

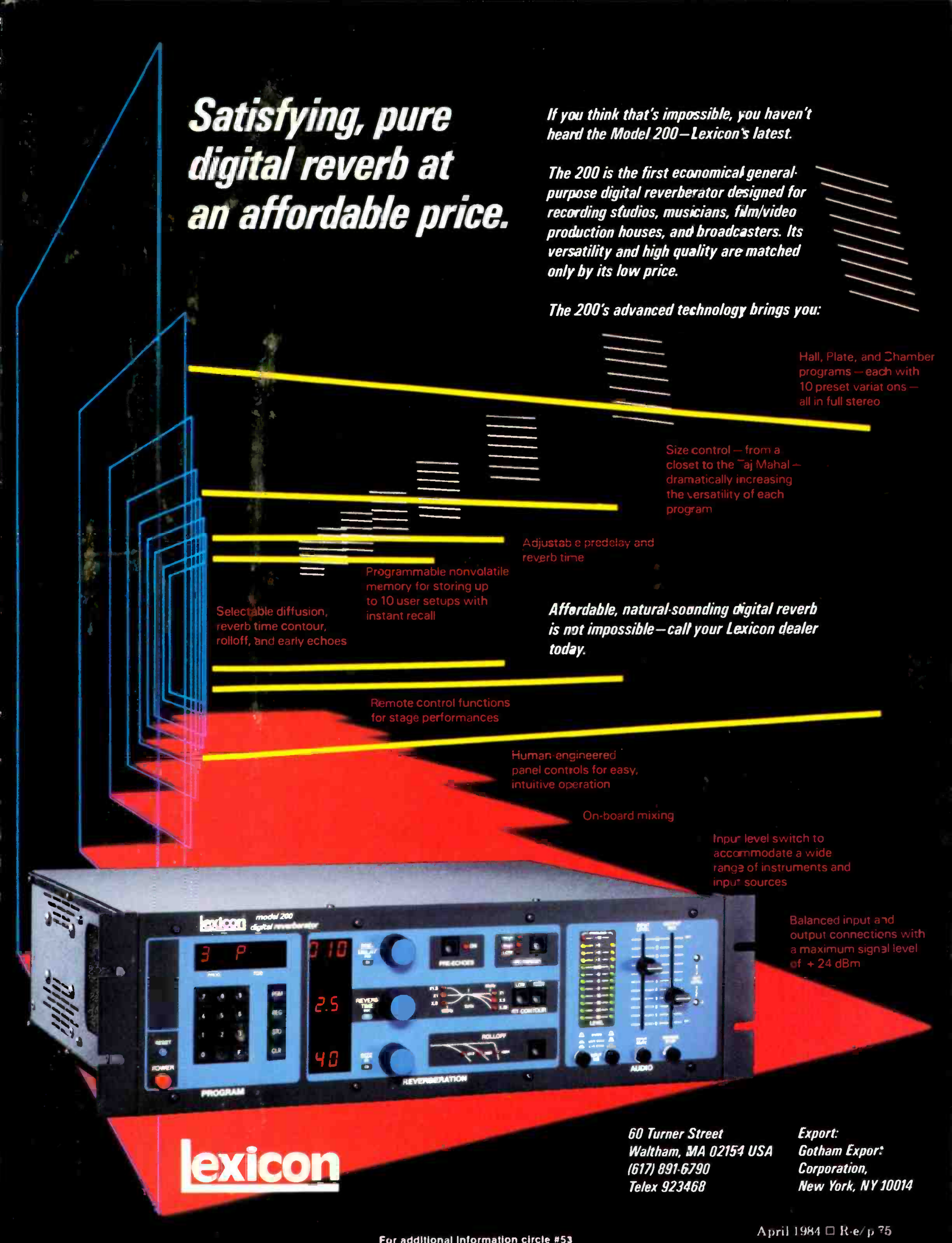
R-e/p: Is there such a thing as the "Tom Anthony sound?" Maybe little hooks that denote your style?

TA: I hope not! I depend on there not being such a thing. I don't want to be pigeon-holed as just a rock-and-roll writer, a country-western writer, or whatever. At *all* costs, I avoid having two jingles sound similar. In fact, the variety is what makes this business so exciting. I love jingles, because I love to write music that is completely different each time.

R-e/p: So you plan on doing this for a while.

TA: Absolutely! I intend to stay in the industry for quite a while, dabble in antique cars [Anthony now owns three], maybe produce a syndicated TV show, and maybe even take a year off, write a book or sit on my ass in the sand somewhere. Who knows? In truth, *nothing* is planned. I live life from day to day. I'm not a planner, because I never know what's going to walk in that front door.

■■■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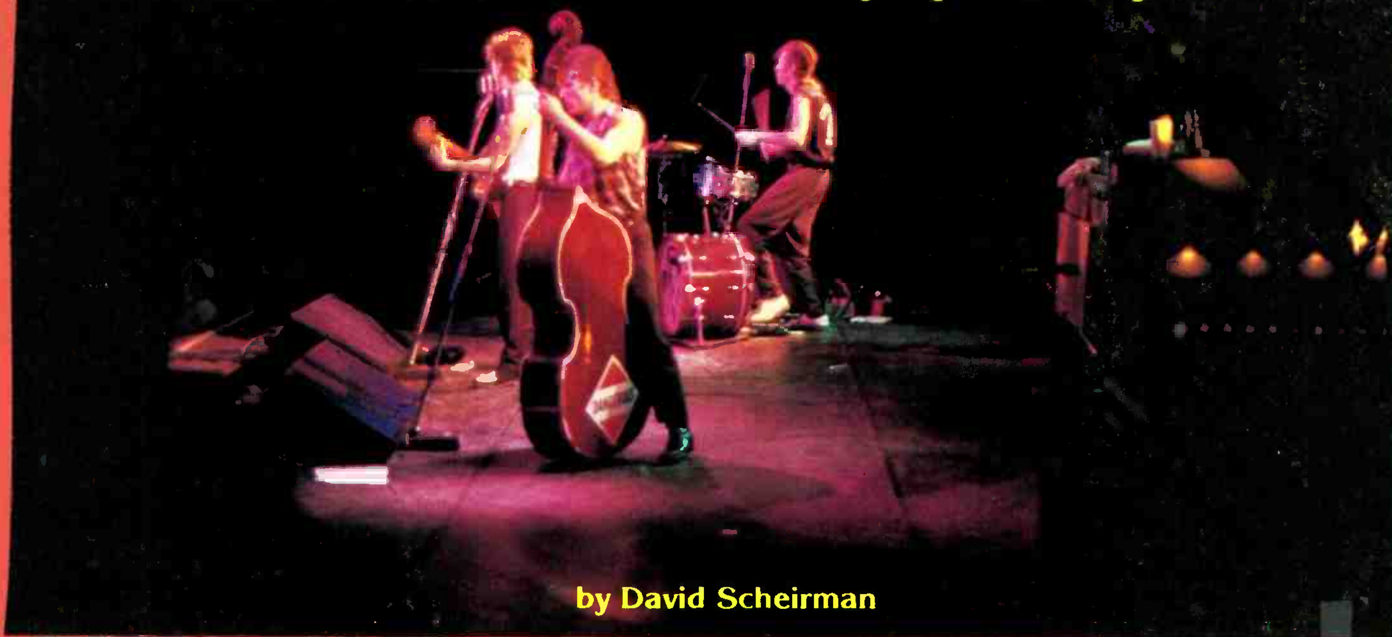
STRAY CATS ON TOUR

With dB Sound's New Four-Way Speaker System



Photography by David Scheirman

by David Scheirman

The Stray Cats, an American trio that first struck Gold in Europe, came to the attention of the listening public in 1982 with the distinctively original sounds of such tunes as "Rock This Town," and "Stray Cat Strut." The band's debut American album (*Built For Speed*) went Platinum, and the band members' black leather jackets, bright tattoos, and rockabilly-influenced music have turned the Cats into a pop phenomenon.

From an audio perspective, the Stray Cats offer an interesting exercise in simplicity. The group's sound is alley-cat lean, with only three performers on stage. Electric guitar, acoustic bass and snare drum back up singer Brian Setzer's Eddie Cochran-influenced vocal stylings. More than just another Fifties revival band, the Stray Cats have created a distinctive sound that is an interesting blend of blues and country-western.

Providing complete concert sound reinforcement services for the group is dB Sound, Inc., of Des Plaines, Illinois. "Wherever the group does shows, we send a system with engineers," relates dB president Harry Witz. "When they first hit in this country, they were playing in clubs. Now, the current tour includes venues all over North America. Sometimes we leave most of the system in the truck, and sometimes we have to send out supplementary gear in the larger arenas. But we have had a basic hardware package out on the road with the group since last year."

To observe the dB sound system first-hand, this writer attended a Stray Cats concert which took place in late 1983 at the Del Mar Fairgrounds in Southern California. Since dB's client list includes just about everyone from Kansas to Kiss, and each road system is slightly different, I also observed several other concerts featuring the company's hardware. As with most large touring PA companies, dB's system components and packaging are in constant evolution. The system out with Stray Cats was a two-box, four-way modular system designed to be either stacked or flown.

House Loudspeaker System

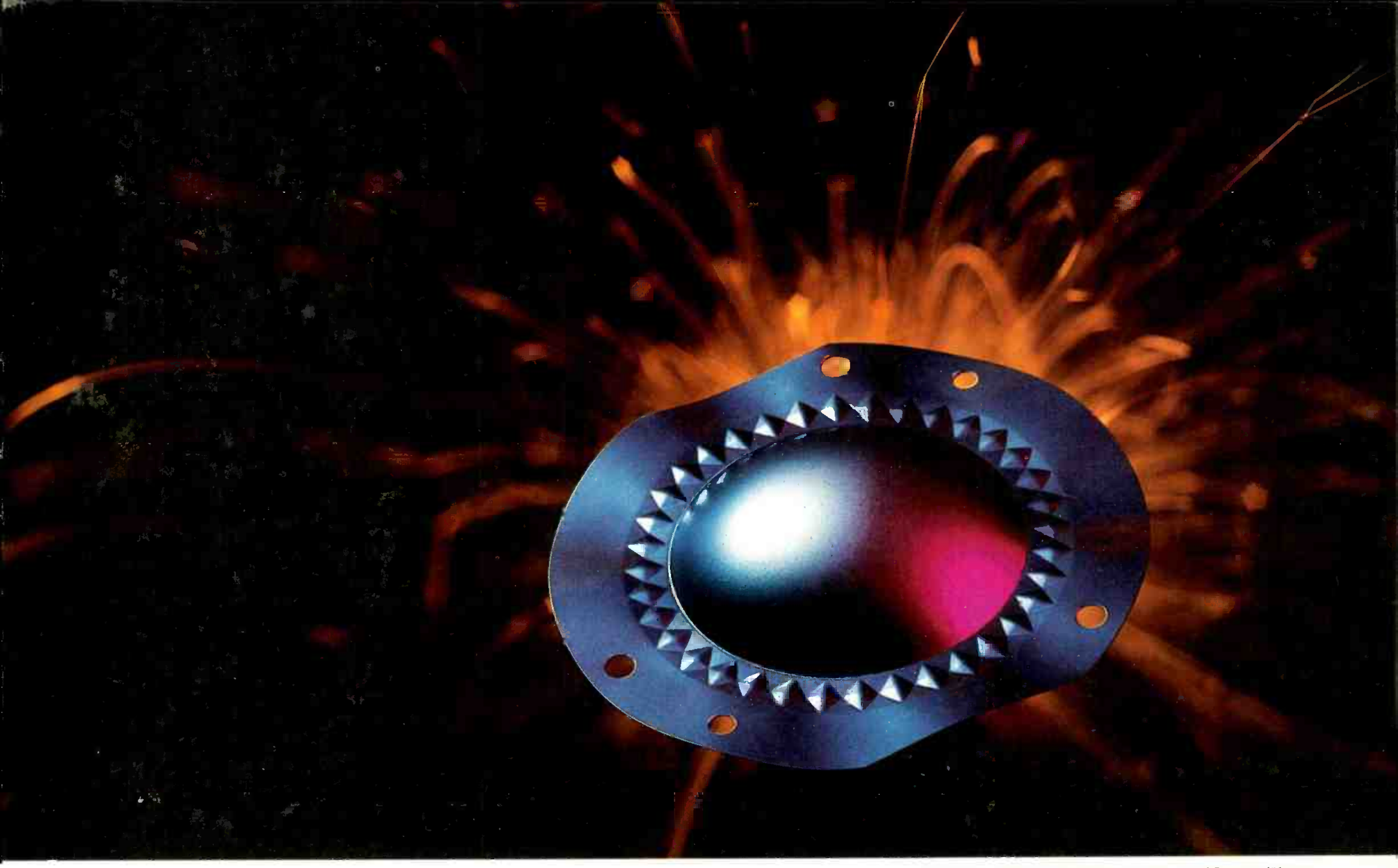
Everyone knows that dB stands for decibel which, among other things, is used to measure perceived loudness. And over the past decade dB's sound systems have had a reputation for being "loud."

"Originally, we used stacks and stacks of folded-horn bass bins and lots of radial horns, just like everyone else did," Witz explains. "And we began to get a reputation for concert systems which were capable of delivering extremely high sound-pressure levels. Over the years, we developed our two-box modular system, which combined the advantages of horn-loading with the portability and ease of transport of the composite boxes. The Stray Cat's system for 1983 represented our standard 'block' system. Over the past year, however, we have been working on a new two-box system which will be smaller and lighter, yet offers increased efficiency and frequency response characteristics." [See accompanying sidebar for description of new four-way system — *Ed.*]

The Stray Cats show at Del Mar Fairgrounds featured a stacked sound system, deployed in an exhibit hall with a capacity of several thousand persons. The show was set up in dance-concert style, with rather narrow sound wings attached to a temporary stage constructed of risers. The dB system was made up of multiples of two types of cabinets: a 4 by 15 bass bin, and a three-way horn-loaded high-frequency cabinet (Figure 1). In larger venues where space permits, the same cabinets are easily hung with chain-motor hoists from steel grids to provide a flying system (Figure 2).

dB's original bass cabinet is a horn-loaded multi-driver chamber. Four 15-inch JBL 2225 loudspeakers are front-loaded at the rear of four separate horn sections, with structural bracing also providing chamber separation. Cones are visible from the front of the cabinet when the protective grillcloth frames are removed. Two bass cabinets and two mid/high cabinets comprise one speaker "block," totalling four cabinets. A single amplifier rack powers each block.

The mid/high cabinet houses another pair of 15-inch speakers for the mid-bass reproduction (Figure 3). The drivers are JBL E-130s (8 ohm) housed in a horn-loaded section. Upper midrange frequencies are passed through a pair of McCauley Model 482 radial horns backed with JBL 2482 drivers. McCauley 421 radials loaded with JBL 2425 compression drivers handle the sys-



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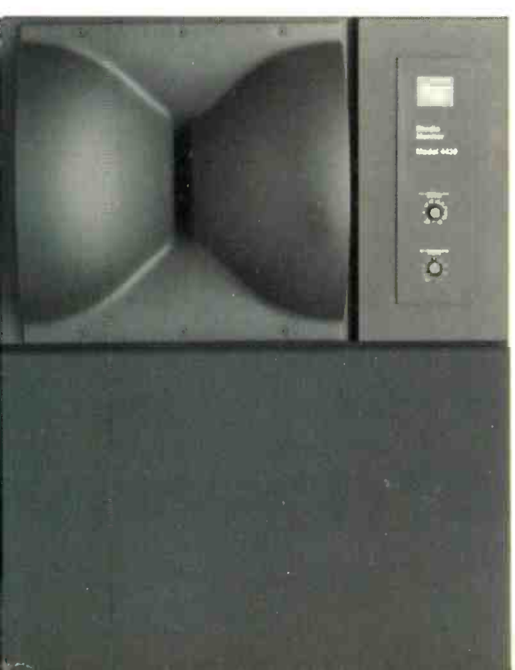

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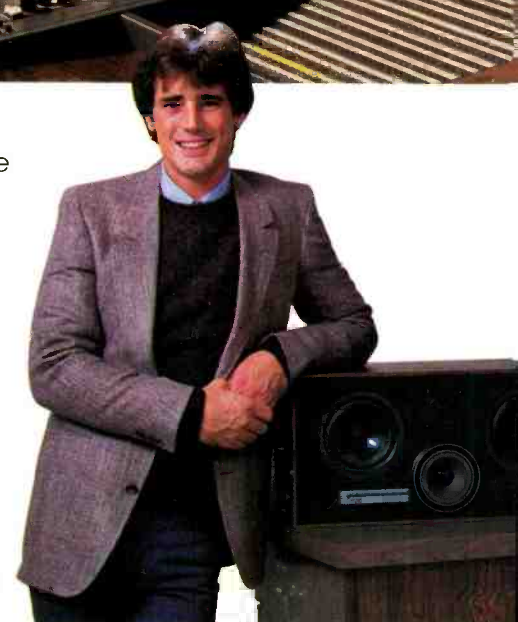

Ampex, 3M, MCI/Sony, Otari, Soundcraft, JBL, U.R.E.I., Westlake Audio, Aphex, AKG, Neumann, Sennheiser, Shure, White, Eventide, Lexicon, Crown, BGW, A.D.R., Yamaha, BTX, Valley People, DBX, Bryston, Studer/ReVox and many other professional lines.

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tem's top end.

"A two-box system such as this one is quite a bit more flexible in terms of setup than a one-box composite system," explains dB monitor engineer Jeff Roeschlein. "And it is a lot easier to travel with, than a system made up of separate components. Of course, a system with separately-boxed components would be easier to stack, because it would be lighter, but I think it definitely goes a lot faster on setup and teardown this way. And your truck pack is much simpler."

JEFF ROESCHLEIN

Four "blocks" per side were available with the Stray Cats system: a total of eight bass bins, and eight, three-way boxes for each sound wing. In the Del Mar Fairgrounds' long, narrow exhibit hall, dB engineers had elected to place 13 boxes per side: six bass and seven mid/high cabinets.

"We don't always have unlimited space when we stack the system," relates dB house engineer Keith Dircks. "A lot of the decision-making process when it comes down to deciding how to stack in a given venue depends on the available space. In here, we were lucky . . . we have at least 20 feet of wing on each side. That's pretty good for such a narrow hall. We have half of the cabinets stacked to utilize their horizontal dispersion, towards the inside. The outer half of each stack, against the walls, is set so that the horn flares are giving more of a vertical dispersion pattern, to keep from sending a lot of acoustical energy right into the walls."

KEITH DIRCKS & HEIN HOVEN

But aren't lower frequencies less directional than high frequencies, even in horn-loaded cabinets? "That's true," Roeschlein states. "And, in a smaller stacked system such as this one, it is not quite so critical to look at your bass-dispersion patterns. But when a full system is assembled for a larger venue, such as an arena, it is *very* important to pay attention to your alignment, your speaker coupling, things like that. You are building a large array which should operate as a point source, *not* just a jumble of boxes."

Power Amplifier Section

With only three stereo amplifiers powering a block of four cabinets (comprising 20 separate loudspeaker components), amplifier units with a relatively high power rating are required. dB's sound systems traditionally have been powered by Crown amplification. "We have tried out a lot of different amps, both in the lab and in the field," Witz explains. "We have found the Crown PSA-2 to be very powerful and extremely reliable, even at two-ohm loads. For our new systems, we have been purchasing Crown's new Delta-Omega™ power unit. It is by far the most powerful commercial amplifier we have ever run tests on."

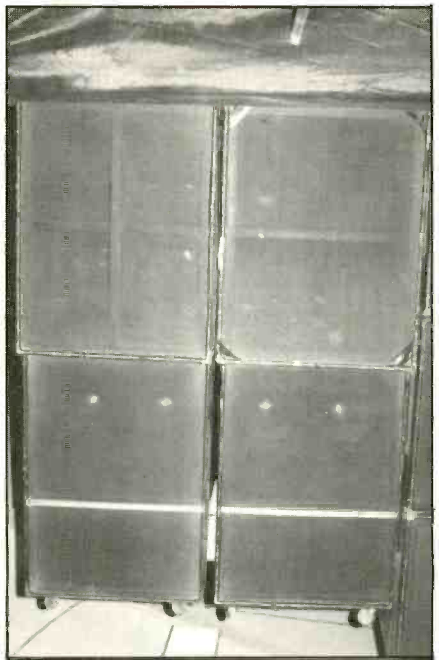


Figure 1: Two-box, four-way loudspeaker system set up on stage-level sound wings. Cabinets shown stacked vertically here are at the outer edge of the speaker stack, only a few feet from the wall.

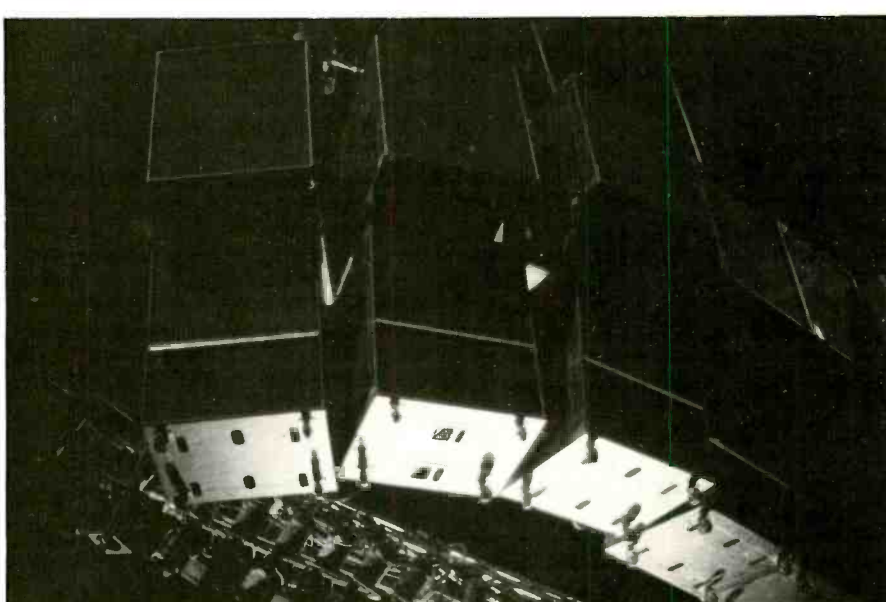
Amplifier racks for the Stray Cats system housed three PSA-2s for an estimated available power rating of 3,600 watts RMS per rack (Figure 4). The bottom unit handled four bass 15s per side at a two-ohm load. Each channel carried two drivers from one cabinet and two from the adjacent one, guaranteeing that if a channel went down, each cabinet would still have half of its low-frequency output. The middle PSA-2 in each rack drove two, 15-inch mid-bass speakers per side, each channel seeing a four-ohm load. The top amplifier was split left and right for upper mids and highs respectively. Each channel drives four compression drivers for a combined load of four ohms.

A junction box at the end of multipair cables delivered left and right output sends from the house drive rack to each sound wing. As the crossover sends entered each amplifier rack, the input signal to each amplifier channel was raised by 50 millivolts with an active line receiver, designed and built by dB Sound. "This circuitry makes up for line loss, and lets each PSA-2 see unbalanced signal," Roeschlein explains. The dB line receiver was contained in a rack-mount panel at the top of each rack. The panel also contains both XLR and ¼-inch phone jack connectors for accessing the bass, mid-bass, upper-mid and high sends, as well as the input side of each amplifier channel.

Stage Sound

"A lot of people ask me what we do to make the band sound the way they do," Roeschlein says. "Really, though, it is a very simple and straightforward setup. The sound of the group is *very* tight to start with, and their on-stage volume is relatively low. They tend to rely on our stage monitoring system rather than huge stacks of amplifiers to hear what they are playing. Brian's guitar amp is really a bit loud for its size — it's an old

Figure 2: The same cabinets are designed to be used in hanging arrays, with integral rigging hardware provided. Here, the bass bins are shown on top, with three-way boxes forming the bottom row.



SOUNDMAN'S NOTES FROM THE ROAD
Andy Chappel's Venue Notes from Across the Country

ST. ANDREWS HALL
431 East Congress
Detroit, MI
(313) 961-8137
Type of Venue: Small Hall
Capacity: 850
Andy's Rating: * * *
Acoustics: Okay when room is full.
House Soundman: Chris Panacki
Mixing Position: House left.
Console: Soundcraft 24×8.
House PA: None. Outside PA company, Chris Panacki Sound (313) 353-6398.

Monitors: None.
Stage: 32×15×4.

House Power: 300 amps.
Crew: Good.
Load In/Out: From street, up about 10 stairs.

Overall View: Good gig, PA sounds good.

Recommendations: Advance gig to make sure you have what you need. Good PA company.

FIRST AVENUE
9 North 7th Street
Minneapolis, MN
(612) 338-8388
Type of Venue: Large club.
Capacity: 1,200
Andy's Rating: * * * *
Acoustics: Good.
House Soundman: Art Welter
Mixing Position: Center of house.
Console: Soundcraft 24×8.
House PA: Southern Thunder Sound, Art (612) 645-9736.
Monitors: Good. PM 1000 24×6, w/third-octaves.
Stage: 36×32
Microphones: Sennheiser, Shure, EV, Beyer
House Power: 200 amps.
Crew: Good.
Load In/Out: Good, double doors and ramp up to stage but inside a garage.
Overall View: Very good gig.

Recommendations: Advance gig, make sure clearance is high enough for your trucks.

LARRY'S HIDEAWAY
121 Carlton Street
Toronto, Canada
(416) 924-5791
Type of Venue: Club
Capacity: 500
Andy's Rating: * *
Acoustics: Good
House Soundman: Ken
Mixing Position: House right.
Console: Kelsy 24×4×2
House PA: JBL boxes, Crown power, okay.

Monitors: Not very good for any kind of level.
Stage: 18 × 12
Microphones: Shure, Sennheiser.
House Power: 100 amps.
Crew: Good
Load In/Out: Short Flight of stairs, not bad.

Overall View: Good Gig, PA sounds fine.

Recommendations: Bring in monitors.

TUTS
959 W. Belmont
Chicago, IL
(312) 929-9158
Type of Venue: Club
Capacity: 700
Andy's Rating: * *
Acoustics: Good
House Soundman: Jane
Mixing Position: Rear of house, right.

Console: 24×2
House PA: Good. Loud and clear.
Monitors: Good. 8 mixes w/third-octaves. 1×15 boxes.
Stage: 22 × 12
Microphones: Shure, EV, Beyer.
House Power: 100 amps.
Crew: Okay, they load you in quick.
Load In/Out: Up a lot of stairs, through double doors.
Overall View: Good club to play; mixing position is strange — not true to the room.
Recommendations: Advance gig; make sure they have what you need.

MABEL'S
613 East Green Street
Champaign, IL
(217) 328-5700
Type of Venue: Club
Capacity: 350
Andy's Rating: * *
Acoustics: Good
House Soundman: Tim Vear
Mixing Position: In front of PA stack, house right.
Console: Tapco 20×4
House PA: Yes, JBL boxes.
Monitors: 2 mixes from house console.
Stage: 24×12, 4-feet high.
Microphones:Shure
House Power: 200 amps.
Crew: Okay
Load In/Out: Hard. Up a large flight of stairs.

Overall View: Good gig once you get in; house PA sounds good.
Recommendations: Advance gig to make sure you get what you need; bring own monitors; get there early for load in.

AGORA
1730 East 24th Street
Cleveland, OH
(216) 696-9400
Type of Venue: Large club.
Capacity: 1,000
Andy's Rating: * * * *
Acoustics: Good; sounds real tight when full.
House Soundman: Bob Smith
Mixing Position: Center of house.

House PA: Yes; Hood Sound.

Monitors: Yes; 8 mixes, very good.
Stage: 32 × 20
Microphones: Sennheiser, Shure, AKG, good selection.
Crew: Very good; one of the best.
Load In/Out: Good loading dock, and onto stage.
Overall View: Very good, production-wise. Sound, lights, and crew are very good.
Recommendations: Just call and advance the gig so they know what you need.

CAIN'S BALLROOM
423 N. Main
Tulsa, OK
(918) 584-2309
Type of Venue: Large Club/Ballroom
Capacity: 1,200
Andy's Rating: * * *
Acoustics: Real live — wood floor and walls; tightens up with people.
House Soundman: Steve Munson
Mixing Position: Rear of house about 120 feet back.
Console: PM1000 32 × 4
House PA: JBL boxes; good.
Monitors: Good; 8 mixes w/third-octaves.
Stage: 30 × 18 foot
Microphones: Shure, EV, Sennheiser.
House Power: 100 amps.
Crew: Good
Load In/Out: Okay; through double door and onto stage.
Overall View: Good gig; don't let sound check confuse you — it will sound totally different at show-time.
Recommendations: Listen from floor at sound check; sounds different in sound booth.

2001 CLUB
3500 Main Street
Amherst, NY
(716) 837-2001
Type of Venue: Large Club/Disco
Capacity: 1,500
Andy's Rating: * * *
Acoustics: Very good.
House Soundman: Larry
Mixing Position: Center of house.

Console: Yamaha PM2000
House PA: None; Local PA company — Systems Inc., (716) 886-7091.

Stage: 25 × 16 foot
Microphones: Shure
House Power: 300 amps.
Crew: Good
Load In/Out: Easy; through large doors right up to stage.
Overall View: Nice club, good PA.

Recommendations: Just advance gig.

SPIZE
1058 Route 110
Farmingdale, L.I., NY
(516) 531-9563
Type of Venue: Club/Disco
Capacity: 750
Andy's Rating: *
Acoustics: Good; dead room.
House Soundman: Alan
Mixing Position: Center of house.

Console: Tapco
House PA: None; outside PA company — Centour Sound, (516) 747-0245.
Monitors: None
Stage: 25 × 16 foot
Microphones: Shure
House Power: Bad
Crew: None
Load In/Out: Easy, through big doors.

Recommendations: Bring in your own generator; advance gig to find the sound you will need. □□□

Note: Andy Chappel's personal rating scheme is based on a maximum score of four stars.

STRAY CATS

Fender tube top and cabinet — but it's certainly putting out nothing like a wall of Marshalls would!

"So you have a group which isn't particularly loud on stage compared to a lot of other rock acts. And I am not reinforcing the drums and bass very much through the monitors . . . it's primarily a vocal-oriented monitor approach. And yet, it gets really loud up here sometimes! The house sound is mixed at a fairly high level. And on some of the smaller stages we play, you may see the guys only 10 or 15 feet away from the stacked bass bins for the house sound

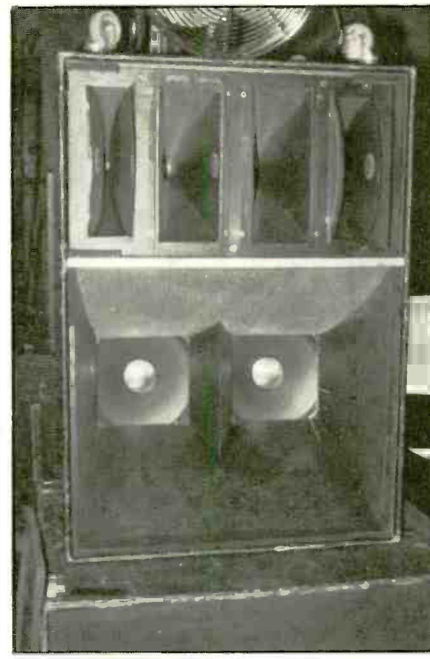


Figure 3: dB Sound's old-style three-way box, housing two JBL E-130s, two 2482 drivers, and two 2425 drivers; horns are manufactured by McCauley.

So I'll maybe get some low-end interference happening up here, 150 cycles, 200 . . . that sort of thing. And that's the sound I'll have to get above with the vocal monitor mixes."

Roeschlein does most shows for this act with a total of seven monitor mixes: stereo sidefills, and three separate floor slant mixes for the performers. "In the larger places, where we hang part of the house sound system," he explains, "I'll also usually fly a pair of overhead monitors, and make them stereo, so that's another couple of mixes. That can be touchy sometimes, though, I hang them

EVOLUTION ON THE ROAD
dB Sound's New Two-box Four-way System

dB Sound's latest design technology for sound systems was developed in conjunction with a custom-tailored computer program. Flares on the company's new HP Series horn-loaded enclosures were designed with assistance from the program, which can accurately predict high-frequency rolloff, the cabinet's efficiency ratio in percentage, and its acoustical output in watts. The program also calculates cross-sectional area at any point of the horn's throat for conical, exponential and hyperbolic flares.

"We have spent a *lot* of hours on the drawing board with this one," relates dB president Harry Witz. "Once we decided to go ahead with a new design, I gave an Apple II all sorts of data, including the specific size of the cabinet, different horn flare rates, that sort of thing. Chris [Cassell, dB technical engineer] puts all of that into one program, so that I could vary any parameter — throat size, for example — and then see how that would affect everything else. Then we took six different bass bins into an outdoor test environment, and checked them all out until we got the computer program set up correctly, with the help of a General Radio calibrated microphone, and a four-color CRT computer-based analyzer. When we got all of the answers to start matching up with our test results, it was obvious to us that we wanted to use horn-loading for the new system."

While the output of the new system is more than double that of the company's already high-level systems, the cabinets have been reduced dimensionally by over 25%. The end result is a modular system built of 36-inch square cabinets that can be combined in various symmetrical configurations to form hanging arrays that are aesthetically pleasing and technically correct. Arrays of up to four blocks (16 cabinets) are easily flown from one grid in a four-by-four fashion.

The new system is finished with a black exterior and dark blue foam grill. All cabling and rigging hardware is contained within the cabinet for quick and easy setup and takedown. Each system block is comprised of two HP-3 three-way cabinets, and two HP-B bass cabinets, containing a total of eight, 15-inch speakers, four, 12-inch speakers, four, two-inch compression drivers, and eight JBL "Bullet" drivers. The block is powered by a combination of two Crown PSA-2s and one of the new Crown Delta-Omega™ 2000 amplifiers, for a total power exceeding 4,800 watts per rack. Initial test results have shown the system to have a 40 Hz to 20 kHz bandwidth, at over 4 dB more output than the company's already powerful systems. An Intersonics servo-driven subwoofer system is also available for additional reinforcement in the 20 to 100 Hz range for special applications.

This new flying system is said to offer compact size, high output, and a wide-ranging frequency response. With truck space at a premium these days, the ability of suppliers such as dB Sound to offer a touring act "more sound with less gear" may give such a company an edge in the highly-competitive concert sound market. □□□

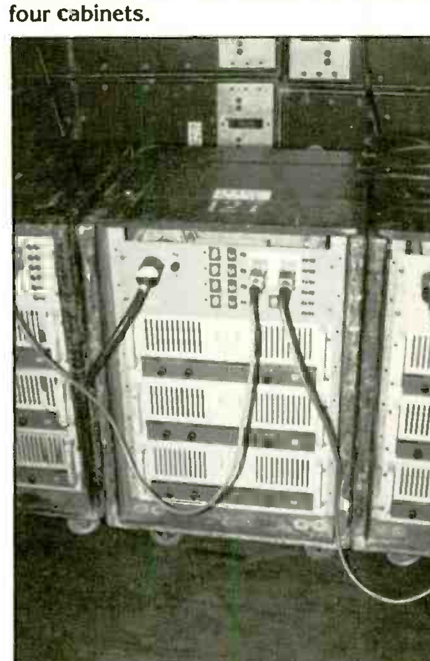
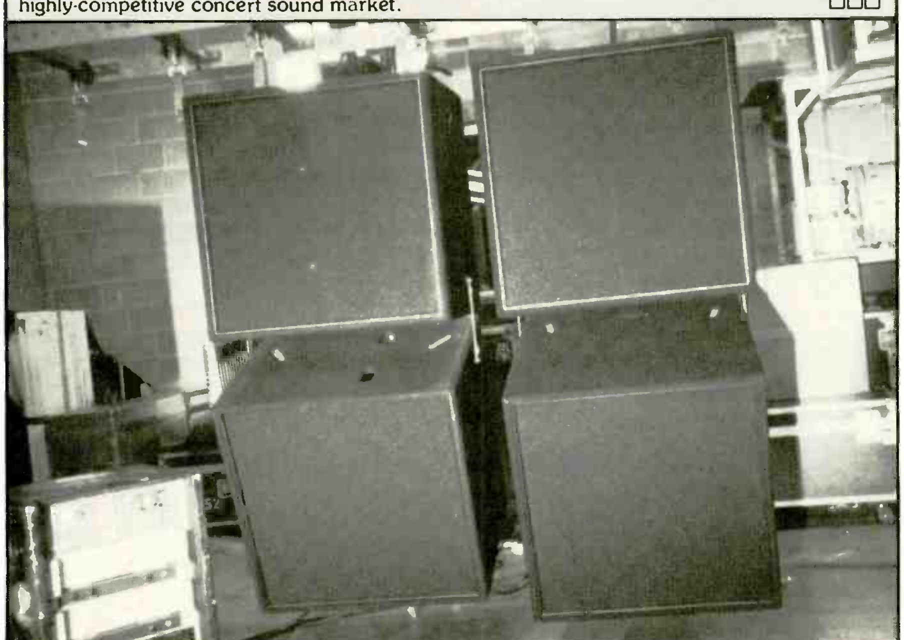


Figure 4: Standard amplifier rack with three Crown PSA-2s driving one block of four cabinets.



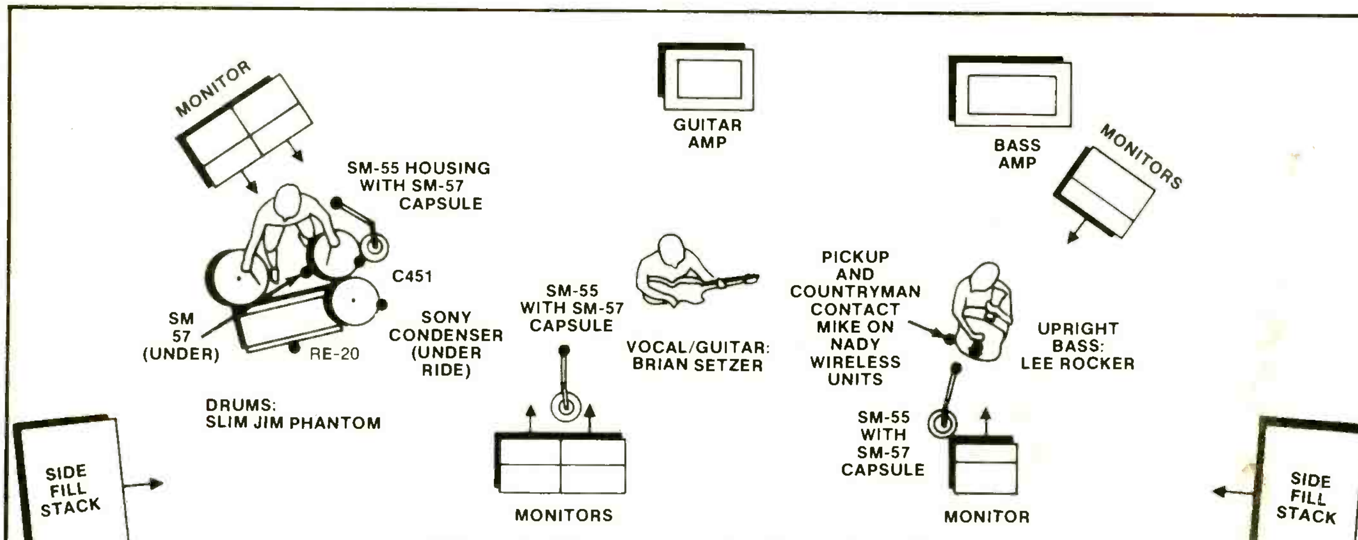


Figure 5: Stage Layout for Stray Cats' Tour.

STRAY CATS

from the front lighting truss. And if it happens to be flown a little higher on a given night, then my overhead mixes are way up there — far enough that there is a slight time delay problem. The sound of the floor slants hits [the band] *before* the overhead sound. And trying to correct it electronically doesn't seem to work well . . . I think that really messes the performers up, to be hearing their monitor sound a bit behind the house mix."

With only three performers — Slim Jim Phantom on drums, Lee Rocker on bass, and guitarist/vocalist Brian Setzer — and three floor slant mixes, the band's sound on stage is fairly clean (Figure 5). "The drummer uses only kick, snare, and cymbal," notes Roeschlein. "And he can hear what's happening around him very well, as he is not putting out a lot of clutter himself. It's a very clean, sparse rhythm section. The bass player gets just a smidgen of his acoustic bass sound from a wedge right behind him; just a bit so he can hear the percussive attack. And he hears himself quite well from the house mix coming back at him. The guitar player's mix is loudest, but I really don't have any problems at all with these guys. The sidefills get a vocal mix, of course, and just a touch of kick and snare for everybody. Actually, you couldn't ask for a cleaner setup. Three guys sure beats 10 or 20!"

Monitor system hardware includes a Midas 24 by 10 console equipped with PR4 input modules. "Sometimes for our opening act I will actually use up to nine mixes, when I add their requirements on top of what the Cats need," Roeschlein continues. "And we have a sax player who comes out for several tunes during the show, so that's a bit of variety. But I basically have plenty of inputs and out-

Figure 6: Monitor electronics racks housing 10 sides of Klark-Teknik graphics, Midas power supply, communications system amplifier, and stereo cassette deck.

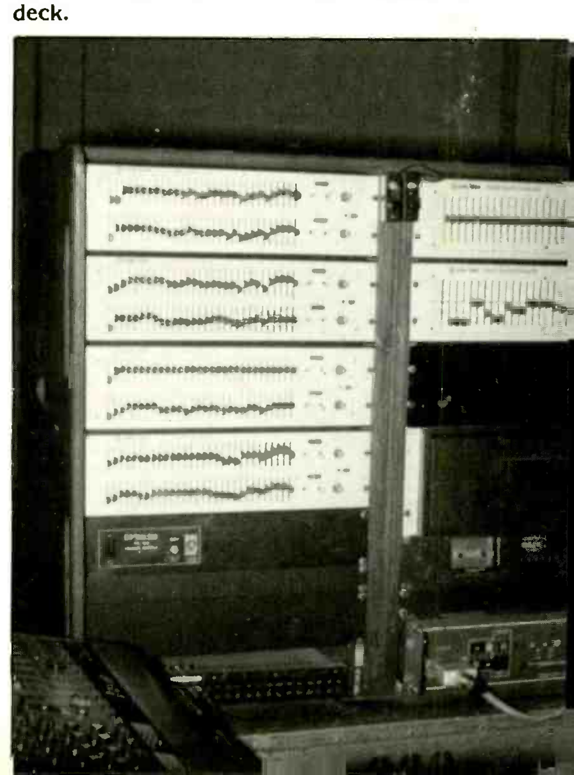


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Figure 7: Low-profile floor slants were paired to give each performer an increased low-frequency presence.

puts with this board, which is a switch from some of the more complex groups that we do."

Klark-Teknik Model 3030 third-octave graphic equalizers were available for each monitor mix output; mixes were bi-amped with Ashly SC-22 electronic crossovers housed in the monitor amp racks. A separate rack contains equalizers and the Midas console power supply (Figure 6).

"The bi-amped floor slant mixes use Crown for power," he explains. "The low-end is handled by DC-300As, and the high-end with D-150s. My sidefills are four-way. In this system we use a Crest 5000 for the low sends, and three Crest 3500s for the upper frequency bands. A UREI 525, four-way stereo crossover splits the signal for me up here. It's a tuneable crossover, which can be quite handy at times, but since my sidefill stacks use the same cabinets as the house system, I have it set for the same crossover points: 400 Hz, 1,500 Hz, and 6 kHz.

"Since the bass cabinet is partially a direct radiator, the low send can be a bit higher than usual. The 15-inch cones are not exactly what you would call front-loaded . . . they are set back in the rear of the horn chambers, and a slot in front of each speaker exposes approximately 50% of the cone area."

Floor slant monitor speakers, of which seven were used for the Stray Cats, comprised a small, compact wedge contaning a single 15-inch JBL speaker, and a two-inch JBL 2441 compression driver mounted on a fiberglass radial horn (Figure 7).

"For this act, a very clean-looking, low-profile wedge is required," Roeschlein explains. "For louder acts, we would use wedges which each contain two 15s. Here, I pair two of them together side by side for an increased presence, so I get the benefit of two 15s without having to set out large, bulky boxes."

Drummer, bassist and guitarist each had a pair of the compact slants, with an additional box placed upstage of the bassist. More cabinets were available for the opening act's vocalists, and the sax man's occasional forays onto the stage.

Microphone Selection

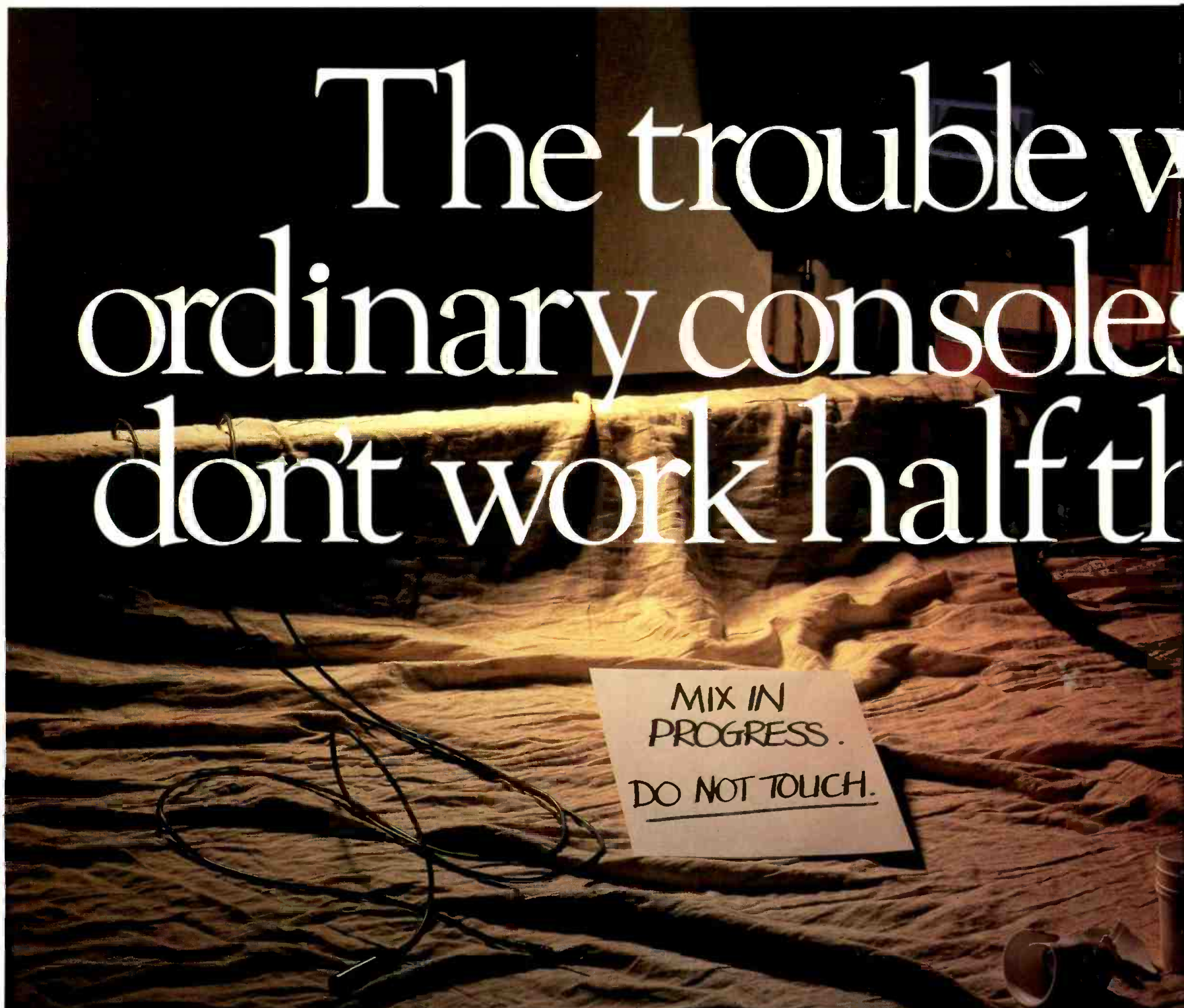
When compared to most contemporary rock acts, the Stray Cats' upright acoustic bass stands out as something out of the ordinary.

"To get the sound of the bass," Roeschlein comments, "we use a regular, heavy-duty humbucking pickup, just like on an electric bass. That gives it the good, rich low bass. Then directly underneath the fretboard on the back side is a Countryman contact mike, which picks up the highs, the percussive effects you get when he snaps a string, that sort of thing. And both pickup and mike are wireless, so you see nothing at all coming out of or sticking out from the bass."

Marc Salter, a member of the Stray Cats crew responsible for tuning the bass and guitars, originally came up with the idea for the upright bass pickup. "I didn't actually do the wiring or the woodwork," he explains, "but I really felt we needed to go with something better than just sticking a mike in front of it, as we'd been doing. We cut a trapdoor into the side body of the instrument, and made a foam-lined compartment which houses the wireless transmitters and the phantom power supply for the contact microphone, along with all the wiring. Before we cut

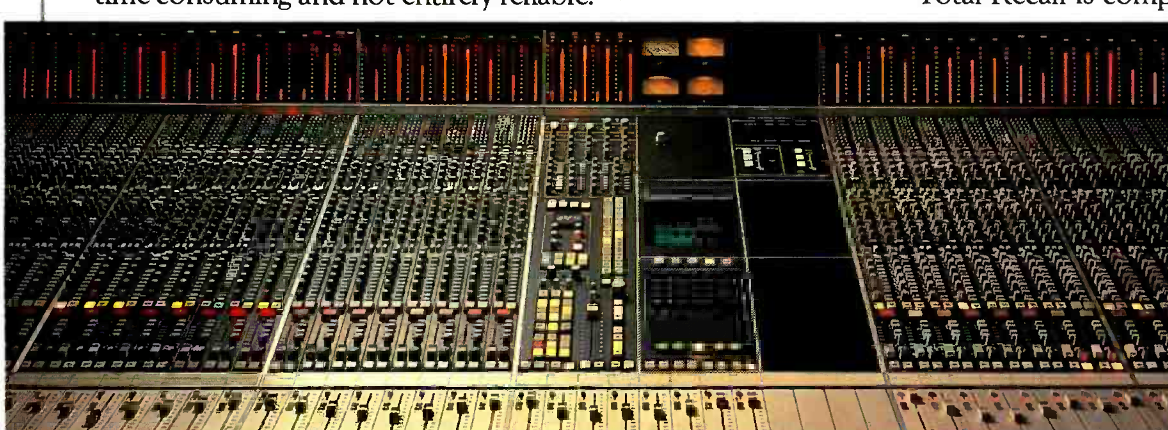
. . . continued on page 88 —

The trouble w ordinary console don't work half th



It's a situation that every studio manager recognises. A client has been in, done some work, and departed to return some time later. Expecting to find the desk as it was left.

Of course, the engineer could always note down all the settings and then reset the desk. But that's extremely time consuming and not entirely reliable.



So, usually, the studio has to stand idle between sessions. Keeping the customer happy, but not keeping the money coming in.

At Solid State Logic, however, we've developed a rather more practical solution to this dilemma. We call it the Total Recall System.

Total Recall is completely independent of all audio paths and allows the console settings to be permanently stored on floppy discs within a few seconds.

So, at the next session it takes only minutes to reload this information, check it on the colour video monitor and return the console to its original settings.

The same thing can be done at the end of each mix

— continued from page 84 . . .

this compartment into the body, it was all quite a mess ... wires and straps and things hanging everywhere. This has cleaned it up considerably.

"We use Nady wireless units, which get fresh batteries every day. You can't really use rechargeable batteries, because they are a bit lower in voltage, and lose their charge too quickly.

"We have probably screwed up the acoustics of the upright bass by installing all of this paraphenalia inside, but we are not really using it as an acoustic bass. There is no need for any sound-absorbing material inside the sound cavity, and feedback is not a problem. It actually plays like a large, oversized electric bass in terms of sound. And we carry a spare bass which has the same electronics compartment built into it.

"To make the sound thing work out, we have the instruments strung with metal-core strings, rather than the gut-type strings which you would see on a traditional upright bass.

"We first decided to set the bass up like this in England, about three years ago. We really had to go wireless, because Lee kept putting the spike at the bottom of the bass through his stage cables about once a week or so! Plus, it was hard for him to spin the bass with everything hanging off it.

Vocal mikes are an interesting hybrid combination (Figure 8). "We have the old classic which the band really likes: a Shure SM-55 housing," explains dB engineer Keith Dircks. "But it has an SM-57 cartridge inside — the modification was done for us at the Shure factory. The on-off switch is not operable, and the mike sounds pretty good, but a bit different."

The minimal drum kit, which consists of kick, snare and cymbal, is miked in a very unobtrusive manner. No mikes or stands are visible to the audience. "We have an ElectroVoice RE-20 tucked inside the kick drum," Dircks continues. "The snare is miked from the bottom with a Shure SM-57 on a clip, and on top with an AKG C-451. That is the mike which Hein [Hoven, the band's house mix engineer] uses primarily for the sound of the snare when Slim Jim Phantom uses brushes. Then, we have a small Sony condenser capsule tucked underneath the ride cymbal, clipped to the cymbal stand. And that's the drum kit!"

House Mix Position

The 1983 Stray Cats tour was provided with a Soundcraft Model 3B house mix console in a 38-in/8-subgroup/stereo-out configuration (Figure 9). For 1984, dB has equipped the system with a new 40-input Midas PR40 board. The company keeps a variety of consoles available in stock, since different engineers have different preferences and

Figure 8: The Shure SM-55, an old standard, is favored for vocals by band members, with one important difference: the microphone has had an SM-57 capsule installed at the Shure factory.

requirements.

The house mix position was supervised by Keith Dircks, with Stray Cats engineer Hein Hoven (an American of Dutch descent now living in London) handling the mix.

"The Soundcraft 3B has been a good board for us," notes Dircks. "There are eight effects sends and eight effects returns. The EQ is four-band sweepable type [plus variable-frequency highpass filter — *Ed*]. Our signal path is as follows: left and right main mix outs hit two Brooke-Siren FDS340 four-way crossovers, after pasing through a pair of Klark-Teknik DN27A graphic equalizers. Each crossover send is sent through its own dbx Model 160 compressor-limiter, so we have eight of those. There is also a spare crossover in the rack, a Soundcraft EX-4S."

A second electronics rack contained the board power supply along with a standby unit, a Roland SRE-555 Chorus Echo, and storage compartments for tapes, patch cables and tools (Figure 10).

The house effects rack contained a dbx frame loaded with four Model 904 noisegates (used on snare, cymbals, and saxophone) and four Model 903 compressors (channel inserts for lead vocal, bass mike, banjo, and kick drum). A TEAC C-3RX cassette deck supplied pre-show music, and an Eventide H949 Harmonizer was present for occasional use on vocals and guitar. A Goldline real-time analyzer, UREI Model 1178 dual peak-limiter, and an Advanced Audio Design D-250 digital delay unit completed the rack. Patch-bay panels on the front of each rack provided easy access to the units.

Mixing The Show

"This group does have a distinctive sound, you could say," remarks house mix engineer Hein Hoven. "And people are all the time asking me what it is that I do special out here. Most of it is the band themselves . . . we have the only upright bass that I know of which has been rigged up to work like a Fender bass. And drum sets without all those tom-toms are practically unheard of. For the bass, I have a direct input from the pickup and a mike in front of the amp to choose from. The trap set has only two cymbals, and usually only one of them is in use at any given time. On the drum sound, I use a Lexicon 224X, set on the constant-density plate — the sound of the snare changes from song to song, but the Lexicon gives the snare sound a bit of continuity. It's on there all the time."

Hoven feels that the dB system has been an excellent choice for the Stray Cats tour. "We play in a lot of very different venues, and the PA has to be *very* adaptable. Small clubs sometimes. Or theatres where we have to hang half the cabinets . . . large arenas . . . outdoor dates. This system has been great for us. And we really won't go anywhere without having Keith [Dircks] along, and Jeff [Roeschlein] on monitors. The crew is great, the gear is great, the support from the shop is great. What more can I say? I am looking forward to the new dB cabinets, which we should get for the next tour." [The Stray Cats tour became the company's first account to utilize the new four-way loudspeaker system described in the accompanying sidebar — *Ed*.]

Concert Sound Assessment

When a touring act makes a long-term

Figure 9: Soundcraft Model 3B house mix console, in a 38-by-8-by-2 configuration.

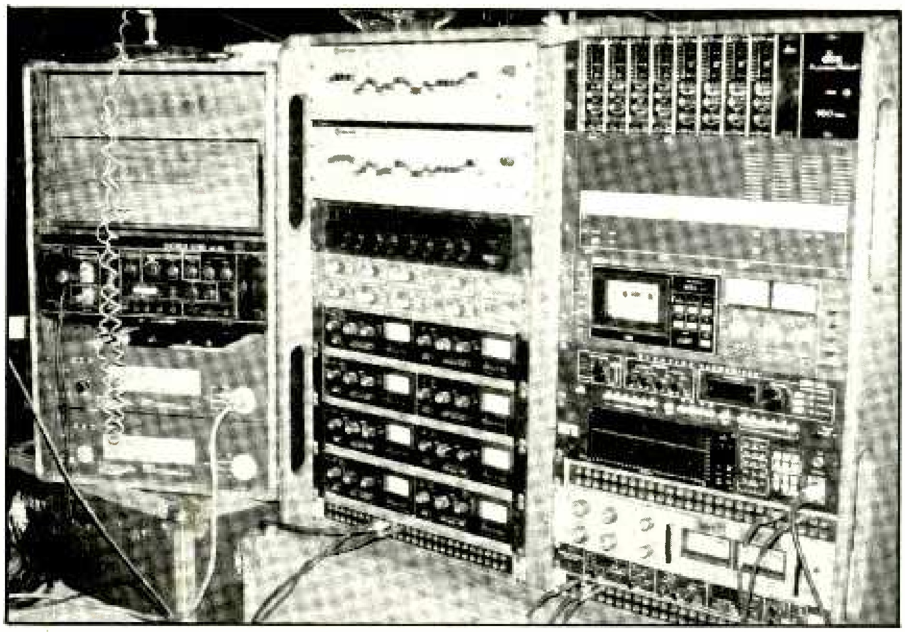


Figure 10: Center house electronics rack contains eight sides of dbx 160 for bass, mid-bass, mid and high sends (left and right). dbx 903s and 904s at right are used as channel inserts on drums, vocals and sax. (Note dual console power supplies and spare crossover.)

commitment to a sound company, there are positive aspects of that situation for both parties: the act comes to count on a consistent crew and familiar hardware, and the sound company gains the benefits of cash flow for research and development.

The Stray Cats show which this writer observed displayed all of these benefits. That act's unique sound was presented to an active and excited audience in a tough acoustical setting. The close cooperation and lack of friction between band and crew members contributed to a good working situation in a dance-concert scene complete with hyperventilating fans that could have had performers and technicians alike pulling out their hair in frustration.

The dB Sound two-box, four-way loudspeaker system is a marked improvement over the old-style typical touring rock system, which used to feature large folded-horn W-bins, and stack of heavy metal horns. The setting in which I heard this system was far from advantageous. (Try to picture an oversized long Quonset hut with a stage at one end, filled with thousands of hyperactive teenagers). However, vocals were strong and clear even at the rear of the facility.

One of the system's strong points was a very full, solid low-end. And no equipment failures, strange hums and buzzes, or other audio-related catastrophes were in evidence. The system seemed to have plenty of brute strength for taking the sound of the show above the very noisy crowd, and it was loud but not unpleasant. This seemed to be one situation where the distinctive "raspy" sound of the mid-range drivers was perhaps advantageous. A more delicate-sounding system might have had trouble asserting itself in a reverberant room where ambient crowd noise levels at times approached 94 decibels. But for the dB Sound crew and gear, it was just another gig on the itinerary.

■■■

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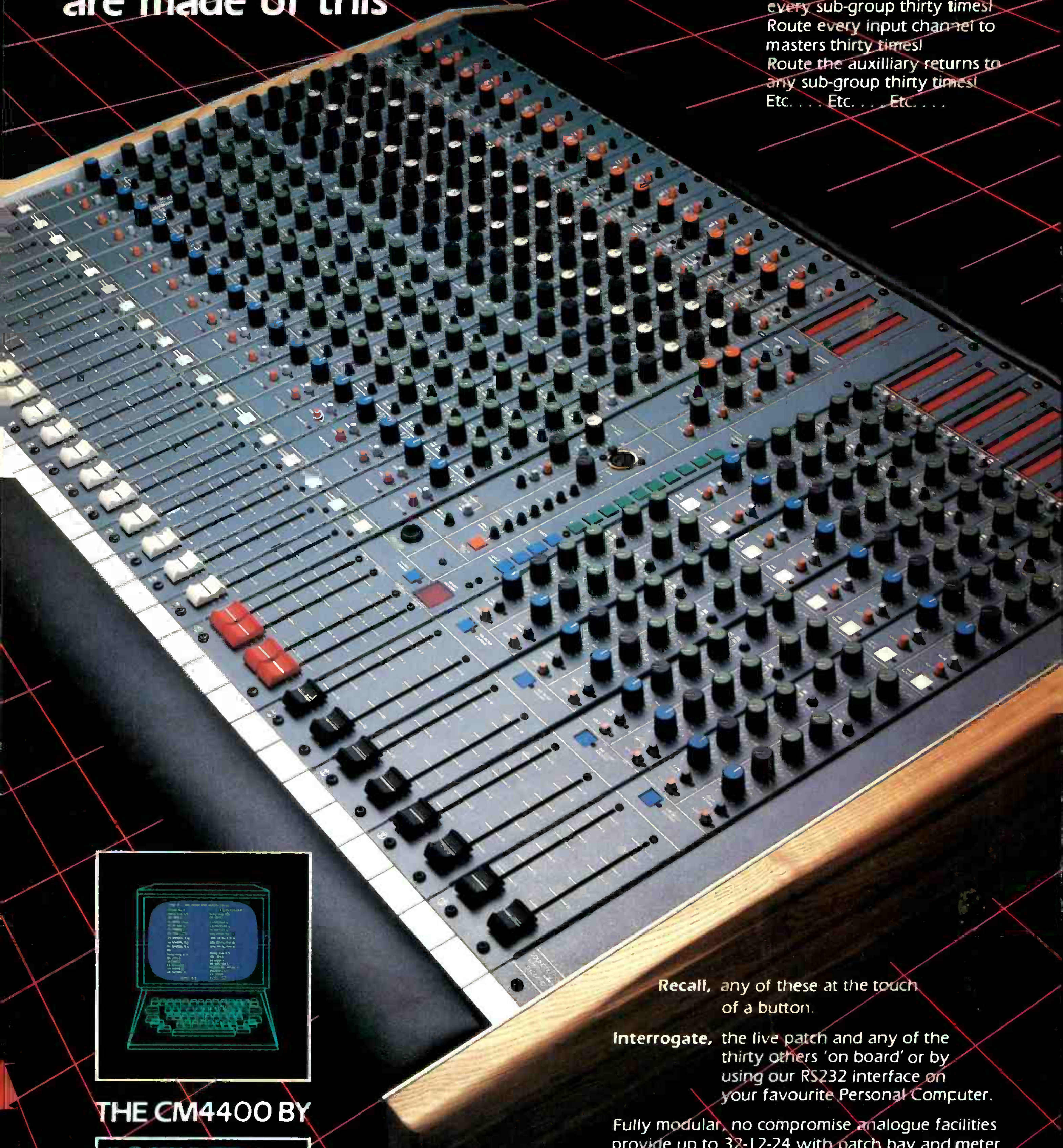

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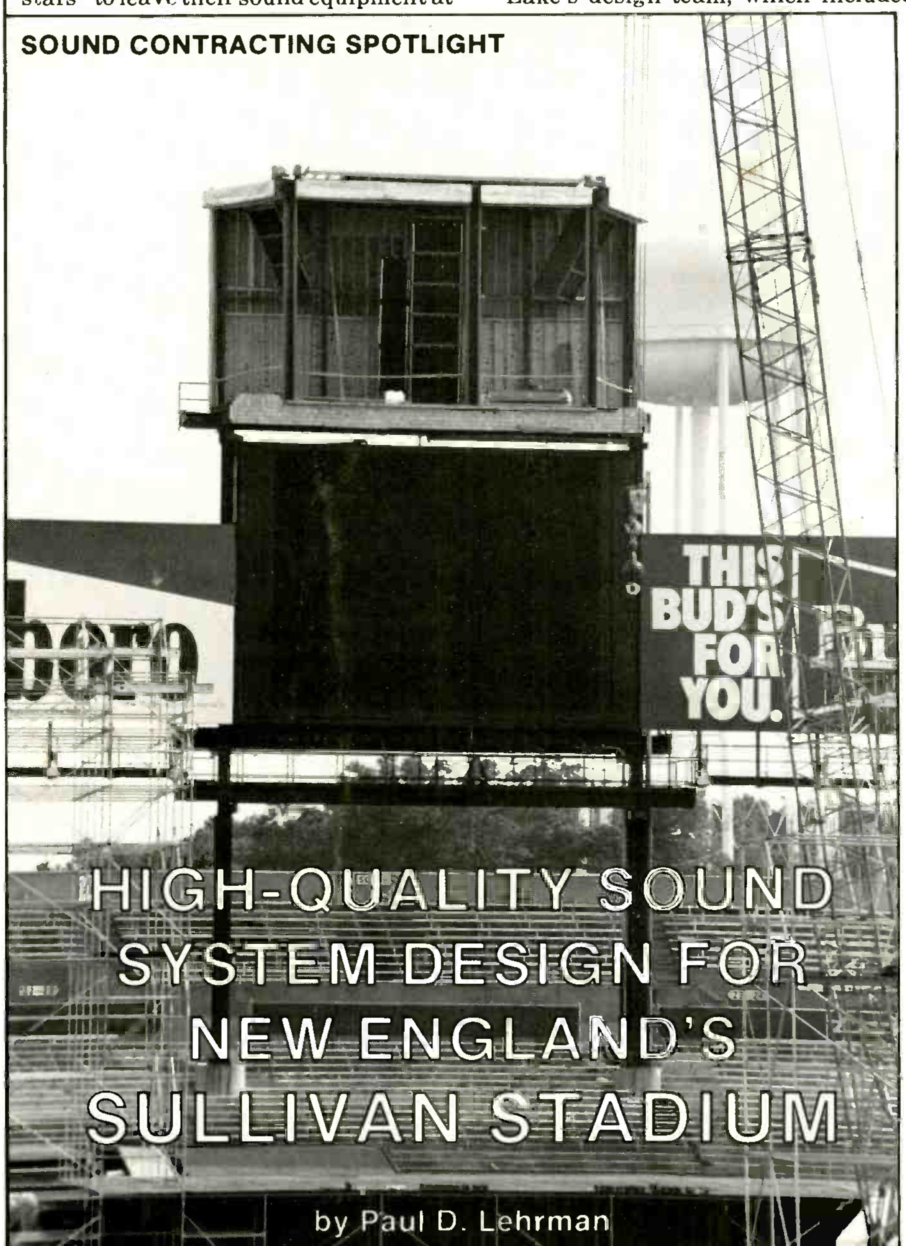
SOUND CONTRACTING SPOTLIGHT

HIGH-QUALITY SOUND SYSTEM DESIGN FOR NEW ENGLAND'S SULLIVAN STADIUM

by Paul D. Lehrman



Photography by Phil Sills/Lake Systems

In sports, the best defense often is a good offense. The New England Patriots football team, sad to say, has neither. But, in terms of overcoming the extremely high noise levels always present at a football game, the recently installed sound and video system at the Patriots' home, Sullivan Stadium, puts up a good fight indeed.

Sullivan Stadium (formerly known as Schaefer Stadium) is located on the outskirts of Foxboro, Massachusetts, a small town about halfway between Boston and Providence, Rhode Island. Besides the Patriots, there's another major attraction in Foxboro — a harness raceway, which immediately adjoins the stadium; a situation which, as we shall discover, has not been especially beneficial to the stadium. Besides football games, Sullivan Stadium is also used for rock concerts and other events. Although the new system may not have been designed to allow "megastars" to leave their sound equipment at home, it *is* capable of putting out large amounts of extremely high-quality audio.

The stadium management decided to upgrade the audio system in 1982, simultaneous with the installation of a new 36- by 20-foot Mitsubishi Diamondvision video screen. At Mitsubishi's suggestion the stadium contacted JBL, who referred the staff to Lake Systems Corporation, a sound/video/security contractor and JBL distributor located in Newton, MA. According to Howard Mullinack, systems director for Lake at the time of the installation, the company's original proposal took only a week to be drawn up. "The system that was in there was about 10 years old," he recalls. "It used Bozak amps and University 'church-bell' horns. Our suggestion to the stadium was to give it all away to a charity they didn't like!"

System Design Brief

Lake's design team, which included Mullinack and systems engineer Dennis Smyers, was asked to develop a system that could handle with equal ease announcements from the booth, calls by the referees on the field, half-time entertainment, and even commercials; one way the stadium generates extra revenue is by "broadcasting" spots to its captive audience for products that cannot be advertised over the air, like liquor and cigarettes.

Architecturally, Sullivan Stadium is basically four huge conical-section concrete slabs laid on the ground at an angle. Behind and underneath the slabs are located the entrance gates, concession stands, and other facilities important to a crowd with a high rate of beer consumption. The stands behind the western end zone are topped off by a four-story, glass-walled structure containing offices and the stadium club and restaurant, and above the stands along the southern side is another glass building enclosing the premium boxes and the press, media, and announce/production booths.

The open area of the stadium — the field and the stands — is covered with a single huge speaker system, suspended 85 feet above the eastern stands. According to Smyers, "The architecture didn't lend itself to a distributed system — there are no overhangs or balconies to conceal speakers. Although," he laughs, "we *did* consider providing headphones for every seat."

The speaker cluster sits in a box atop the Diamondvision screen, supported by six vertical steel girders planted in the stands. The box is trapezoidal in shape, the front wall being 30 feet wide and the rear 40 feet, and it stands 20 feet high and 12 feet deep. The cluster itself contains a total of 88 JBL speaker systems, and weighs in at 27 tons.

Speaker arrays are carefully aimed so as to minimize sound reflections and echoes off the glass buildings. Three sets of midrange horns provide different throw lengths and, according to Smyers, the 6 dB off-axis points of the long- and medium-throw horns coincide with the bottoms of the buildings. Two methods were used to determine the proper aim: a geometric construction technique in which the designers took scaled blueprints of the stadium projected in plan, elevation, and section, and simultaneously drew SPL contours on them all; and a mathematical model that analyzed the space on an HP-41CV computer in terms of cones, and projected theoretical dispersion rectangles within them. "The two techniques usually correlated to within one degree," Smyers offers. "When they didn't, we'd go back and invariably find that we did something wrong on *one* of them."

Highs are handled by 54 Model 2343 horns fitted with 2425J drivers. Model 2482 drivers are used for long-throw coverage, feeding 12 2366 horns for long-throw coverage, six 2365 horns for medium-throw, and four 2360 horns for short-throw. Bass is provided by 12 Model 4550 low-frequency bins, six of which (weighing 1,437 pounds and known collectively as "the Big Woof") are pointing straight out, while three each aim left and right.

Climatic Restraints

Of course, under such conditions there's no way to completely eliminate sound reflection and echoes. The situation is made more complicated at Sullivan Stadium by a factor that indoor sound designers generally do not have to contend with: the weather. "There is sometimes a temperature difference between the field and the altitude where the cluster is," says Smyers. "When the upper air is colder, there's a kind of 'skip' phenomenon in which the sound curves up, and then we get more echo. We didn't realize this would happen until after the system was put in, but it varies so much that there was no way we could have designed it into the system anyway.

"The temperature also affects the frequency response," he notes, "as does the humidity. We get optimum high-frequency response on a cold, damp day. We did consider at one point putting in some kind of humidity-variable high-frequency attenuator, but it wasn't practical. Trying to do any kind of overall equalization would be meaningless except at a very gross level. If you walk around the stadium with an analyzer, or even just listening to the system, the frequency response is always changing, especially when the wind is blowing."

New England is hardly known for its balmy winters, a factor which forced Lake Systems to take extraordinary measures to protect the speaker components. The cluster contains a set of electric baseboard heaters to keep them from freezing. The entire unit is protected when not in use by a motorized set of insulated galvanized-steel "garage doors" —unfortunately, they tend to get stuck occasionally, and so every time the doors are opened or closed, an operator has to climb up a permanently-installed ladder to the cluster and operate them manually.

But the system obviously doesn't sound very good with the doors closed and, since (like the postman) rain, snow, or gloom of fans shall not prevent a football game from reaching its appointed conclusion, the speakers have to be prepared to take all sorts of natural abuse on a fairly regular basis. To prevent humidity and rainwater from collecting in or on the elements, the woofer cabinets are covered with fiberglass, while the horns are treated with Scotchgard repellent, and covered with a special foam with a high "reticulation factor" — that is, it deflects water downwards.

"We had a temporary system installed on top of the stadium club for a while," says Smyers, "and we got serious icing and rusting problems on the hardware." As a result, before the permanent system was installed, all of the driver hardware was replaced with stainless steel, and treated with silicone. Fine nylon mesh was stretched across the throats of the midrange drivers to keep moisture out. "JBL recommended we use Saran Wrap for that," he laughs. "It lasted until we turned the system on!"

Power Amps and Processing

Before the audio signal arrives at the speakers, it is compressed with a UREI LA-4, and then run through a Crown VFX-2 crossover, which splits it into three bands: below 500 Hz, 500 Hz to 4 kHz, and above 4 kHz. Each band is then limited further with a SpectraSonics Model 610.

Power for the system is provided by banks of Crown PSA-2 amplifiers located in a concrete "vault" underneath the stands. There are 28 PSA-2s in all, plus a spare, some of which have their outputs strapped into mono, so that the total potential power is 37,400 watts. "We're setting the amps' AGCs to deliver power levels at half of JBL's published specs," Smyers says, "which seems safe. We even had to lower the amp levels a bit after installation, because it turned out the speaker impedances were not the same as we had calculated them to be."

Central Speaker Cluster during construction. Shown at top of the page are low-frequency side-fill cabinets, and below the front-facing center low-frequency and mid-frequency cabinets for the stadium's North End.

Within the amplifier room and inside the cluster, 12-gauge stranded wire is used for hookup, while 10-gauge wire runs between the vault and the speaker box.

The head end of the system — located in the announce/production booth, where over a dozen audio and video technicians rub elbows and more during a game — is designed for flexibility, but uses a minimum of audio processing. A cassette machine, and stereo and mono broadcast cartridge decks provide music for the cheerleaders to do their thing to. "Whatever the girls bring in for music," says Smyers, "we can play."

An E-V Tapco C-12 Series II mixer handles the music, as well as inputs from the announcer's microphones and a wireless mike worn by the referee. On its way to the PA system, this last is run through an Eventide H910 Harmonizer, set for a small amount of pitch shift, and a UREI Model 539 third-octave filter with a smooth bass rolloff, to reduce feedback and improve intelligibility. "It's overkill," Smyers admits, "but it works."

The announce station connects via a private intercom system with the field spotters and timekeepers. An extensive switching and distribution network allows all input sources to be routed to all outputs — which include a separate PA system covering the areas behind and underneath the stands, as well as

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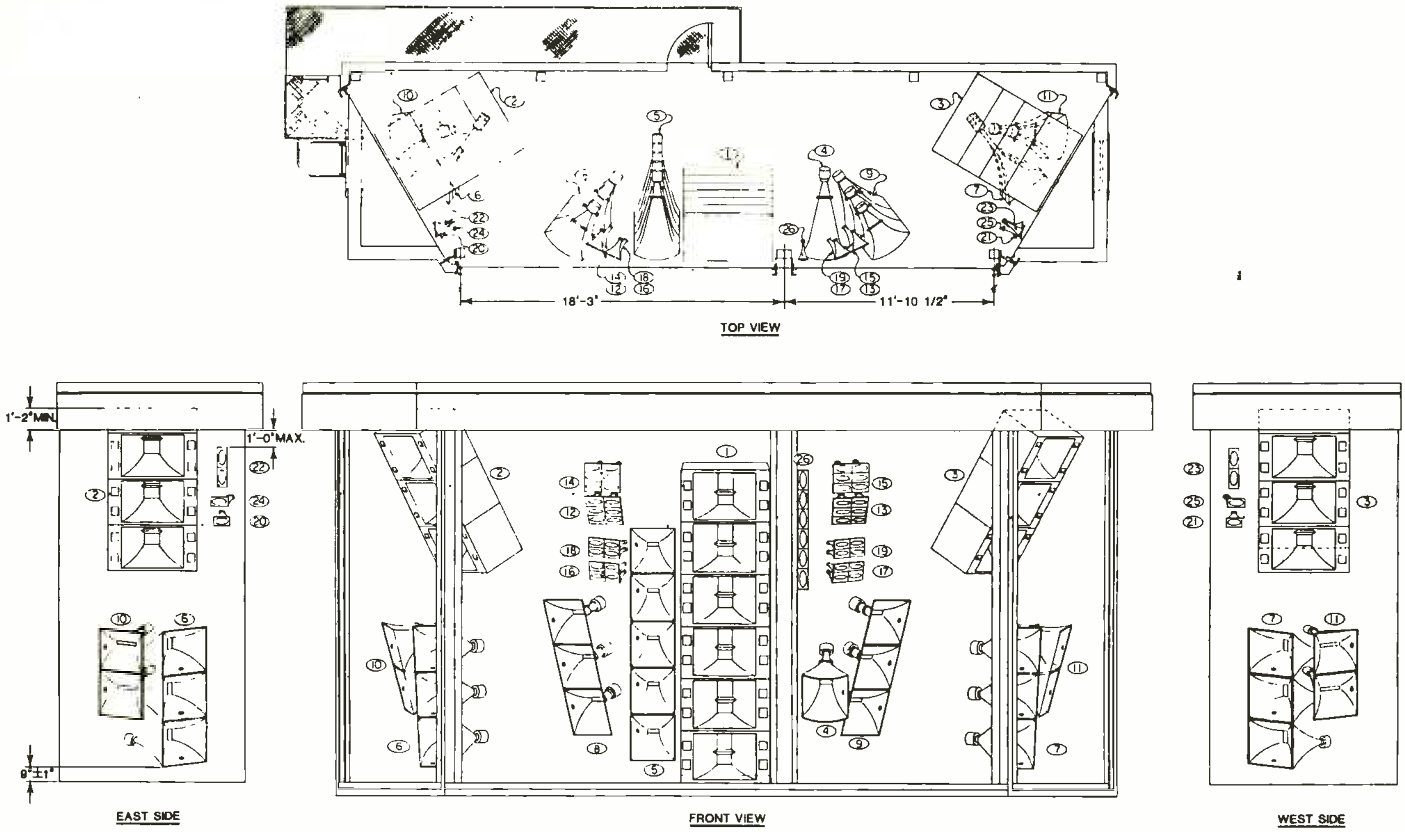


CENTRAL CLUSTER DETAIL

(See Mounting Schedule below right.)

NOTE:
DOT ON HORN DENOTES HORN ORIENTATION

local and network radio and TV feeds, both incoming and outgoing. A special foldback system is being designed that will allow the cheerleaders to hear their music better.

By contrast, the video setup, which was provided by Shintron, would do a small television station proud. There are three ¾-inch JVC U-Matic decks and editor; a copy camera; an ADDA frame store with a capacity for 150 player close-ups; a Sony Dynamic Motion controller for slow-motion replays; Mitsubishi and Quantefont character generators; and a Shintron switcher. All of this equipment, mind you, is used exclusively to feed the 38, 528 individual CRTs that make up the Diamondvision screen; outgoing broadcast production is handled by the broadcasters themselves, from mobile facilities in the parking lot.

The concession- and gate-area sound systems utilize Electro-Voice FC-100 horns powered by a pair of BGW 750 amplifiers. "The amps can drive 70-volt systems without an output transformer, which improves the sound quality considerably," Smyers offers. "We're loading each of them with about 650 watts." Each area is also wired so that it can be paged locally from the announcer's booth.

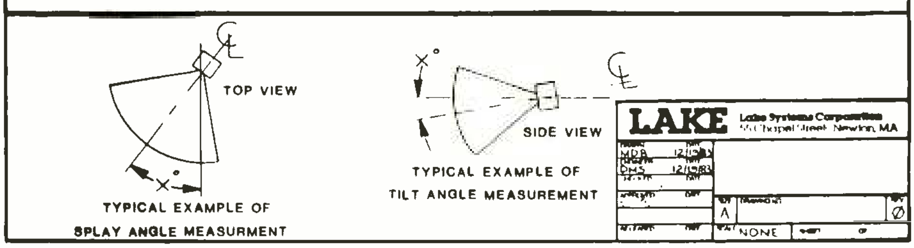
System Compromise

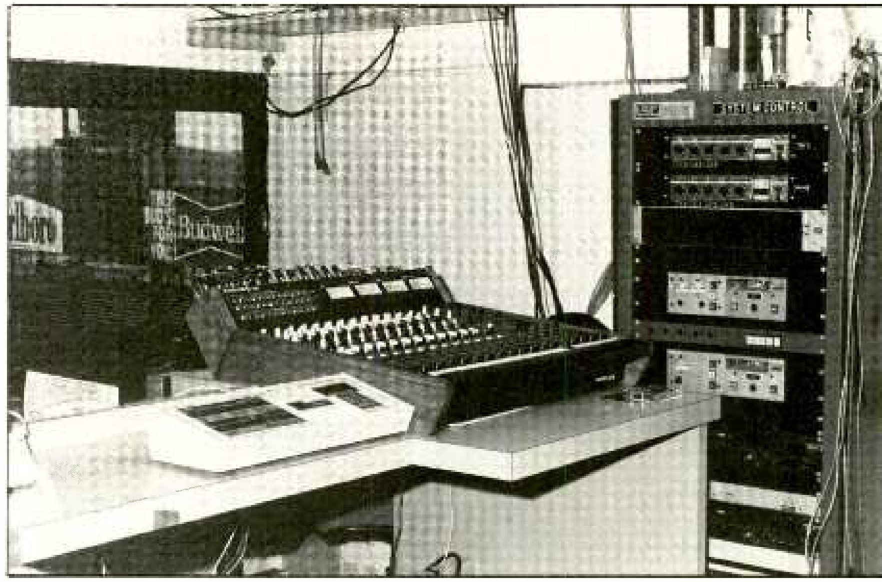
E-V FC-100s are also currently being installed to overcome the one serious problem remaining with Sullivan Stadium's sound system: the quality of the sound in the 20 or so rows of seats directly below and behind the speaker system. As Smyers admits, "It's mud." How this situation occurred in the first place deserves a little explanation. The boundary line between the stadium property and that of the neighboring racetrack lies a few yards east of the stands containing the speaker tower. The stadium architect's original intent, logically enough, was to put the tower outside of the stadium proper. Unfortunately, the huge cranes needed to install the tower supports would have had to take up temporary residence on racetrack property — and the owners of the racetrack, who reportedly have never been happy about the stadium's existence in the first place, wouldn't allow such a thing.

Another factor was that placing the supports behind the stands would have

LOUDSPEAKER MOUNTING SCHEDULE

NUMBER	TYPE	QTY	LOCATION	DEVICES	ARRAY	DEVICE ORIENTATION	DIST. (HxV)	SPL	SPLAY	TILT	AMPS
1	LF	1	CENTER	6-4550	1H 6V	HORIZONTAL	70x10	104-92	0°	8°	1-6
2, 3	LF	2	SIDE	3-4550	1H 3V	HORIZONTAL	70x25	104-87	60°	30°	7-12
4	MF	1	FIELD	1-2366	1H 1V	20H 40V	26x45	99-92	0°	31°	1/2-13
5	MF	1	NO END	5-2366	1H 3V	20H 40V	26x15	99-95	0°	11°	1/2-13 & 14
6, 7	MF	2	FAR SIDE	3-2366	1H 3V	40H 20V	45x9	98-88	36°	5°	15 & 16
8, 9	MF	2	BOX SEATS	3-2365	1H 3V	40H 60V	42x23	103-82	36°	20°	17 & 18
10, 11	MF	2	NEAR	2-2360	1H 2V	40H 90V	46x47	103-80	82°	23°	19
12, 13	HF	2	BOX FAR	6-2343	2H 3V	60H 30V	15x7	95-85	25°	17°	20 & 21
14, 15	HF	2	FAR SIDE	6-2343	2H 3V	60H 30V	15x7	90-78	25°	10°	22 & 23
16, 17	HF	2	BOX NEAR	4-2343	2H 2V	60H 30V	15x10	96-87	40°	17°	24
18, 19	HF	2	MID SIDE	4-2343	2H 2V	60H 30V	15x10	85-76	40°	6°	25
20, 21	HF	2	NEAR LOW	1-2343	1H 1V	60H 30V	30x20	90-80	62°	24°	1/2-27
22, 23	HF	2	NEAR MID	2-2343	1H 2V	30H 60V	20x15	94-76	57°	8°	26
24, 25	HF	2	NEAR HIGH	1-2343	1H 1V	60H 30V	30x20	88-75	82°	10°	1/2-27
26	HF	1	NO. END ZONE	6-2343	1H 6V	30H 60V	20x5	85-80	0°	8°	28





The Audio Control Area features an E-V Tapco Series II console. Signal processors include an Eventide Harmonizer and a UREI Model 539 LF rolloff filter.

required a variance from the local zoning board — Foxboro town regulations do not allow any large structures to be erected within 50 feet of a property line. This problem, unfortunately, was not discovered until after the system was ordered. "They were about to take delivery of the Diamondvision screen," says Smyers, "and that's not something you can store in your basement — they had to put it *somewhere*."

The differences with the town are being worked out, and plans being made to move the audio/video tower to a place where all of the fans can benefit from it — and at the same time get the tower supports out of the way of the view from the cheap seats. In the meantime, a cluster of E-V horns is being mounted underneath the screen, pointed down and backwards.

Problems with the neighbors have not been the only obstacles that the stadium and Lake have run into. Although all of the production equipment was up and running in September 1982, a few months after the contracts were let, it took a year for the rest of the system to go on line. A lot of the delay can be associated with the NFL players' strike.

"Actually, there was no reason why we couldn't go in and work during the strike," says Smyers. "But, as it turned out, it was the best thing that could have happened. In some ways, we really weren't ready. The strike took the pressure off of the stadium to get the system in, which gave us more time for research and engineering.

"For instance, we had originally ordered a different model of high-frequency horn. We brought it in and tested it, and the dispersion and throw characteristics were terrible, so we had to get different ones. We also used the time to build subassemblies, weatherize the speakers, and test everything."

A certain amount of the time was also spent hassling with the stadium architect. "He wanted to make the speaker box smaller to cut down on wind loading, so that he could make the supports lighter," says Smyers. "But we didn't want to do it because it would have affected the dispersion badly. We sent a lot of drawings back and forth."

The Acid Test

Now that all is said and done, the system works very nicely, thank you. The system is reported to be capable of delivering 143 dB SPL at a theoretical single point right at the speaker cluster, while at the furthest reaches of the stadium it still reaches a respectable 93 dB. Electrically, the system is designed to be flat from 50 Hz to 15 kHz, although the speakers roll off the extreme ends, reaching -6 dB at 50 Hz and 10 kHz. Frequency response is fairly even throughout the entire stadium, although it tends to be best on the field and in the high-priced box seats down near the 50-yard line.

In fact, the sound is still pretty impressive several *miles* away. The terrain surrounding Sullivan Stadium is quite flat, and the sound from the system, especially if the wind is right, carries rather well into one of the neighboring towns.

"The stadium forms a huge low-frequency horn aimed right at the town of South Walpole," says Dennis Smyers. He notes that whenever he comes out to the stadium to test the system, the 'phones in the booth start ringing — the first call is usually from the office of the stadium manager; the second is from the staff at the stadium club; and the third is invariably from a resident of the town, 3½ miles away! That's mighty fine coverage for what Lake field engineer Chris Monck likes to refer to as "The world's largest television set, with the world's largest four-inch speaker."

■■■

One of the unwritten laws of today's recording industry is that "Thou shalt continually upgrade your recording gear." Yet eventually a limit is reached where something better isn't readily available, and the only alternative that allows improvement is to "Build it Yourself."

Hollywood-based Sunset Sound Studios, which has hosted such recording legends as the Doors, the Doobie Brothers, Janis Joplin, Barbra Streisand, and Van Halen, has maintained a singular philosophy over the last 20 years — to design and build its own mixing consoles. Paul Camarata, owner and president of Sunset Sound and Sunset Sound Factory, explains why: "Basically, we've never been able to find a quality stock board that would fit our clients' needs. Believe me, if there was a console available that I felt would work for us, we *definitely* would have bought it. We could have saved a *lot* of money!"

Since 1979, Sunset Sound's subsidiary manufacturing company, Sunset Industries, has been focusing its efforts on the development of a "Superconsole," which was scheduled to be installed, debugged and in full operation in the newly renovated Studio One by late March '84. To head the project, Camarata hired an experienced audio design engineer who, he says, "designed everything right down to the wirenut." Don Wolford, who worked on the existing custom console presently gracing the control room of Sunset's Studio Two, handled overall design of the audio circuit and physical layout of the new board for Studio One. Jeff Taylor, who initially was brought in by the studio to assist Wolford with the upgrading of another desk in Studio Three, stayed on to develop the digital logic system that provides the latest custom console with its powerful capabilities. In addition, Larry Weisbech and Mark Sachett handled the actual assembly of module circuitry.

Except for a nine-month period during which their attentions were turned toward remodeling the existing board in Studio Three, Wolford and Taylor have worked non-stop on building the new console for Studio One. Although the process sometimes involved modifying existing equipment from other companies — including API Model 550A EQ modules manufactured by Datatronix, and a Neve NECAM fader automation system — it entailed devising circuits, logic systems, layouts, frame, and everything else from scratch.

CUSTOM CONSOLE DESIGN

Operational Features of Sunset Sound's New Logic-Controlled Console

by Robert Carr

Photography by Murray Kunis

"Sunset's philosophy is to produce something that isn't going to be obsolete too soon." Taylor says. "And we encouraged them to do that. But, because we had only a couple of people working on the design and construction, it's taken almost 3½ years to put it all together. We knew that if we didn't shoot for the ultimate in state-of-the-art, it wouldn't fly by the time we got on the air."

Although to date exact figures haven't been tabulated, total cost of the custom console is rumored to be approaching $750,000. Much of the expense can be attributed to extensive research and development, and the resultant volumes of comprehensive documentation. As Taylor points out, "The only way this kind of expenditure can be justified is to end up with a substantial amount of documentation, comprising assembly pictorials, art work, readouts of stress tests, experimental circuit designs, etc.

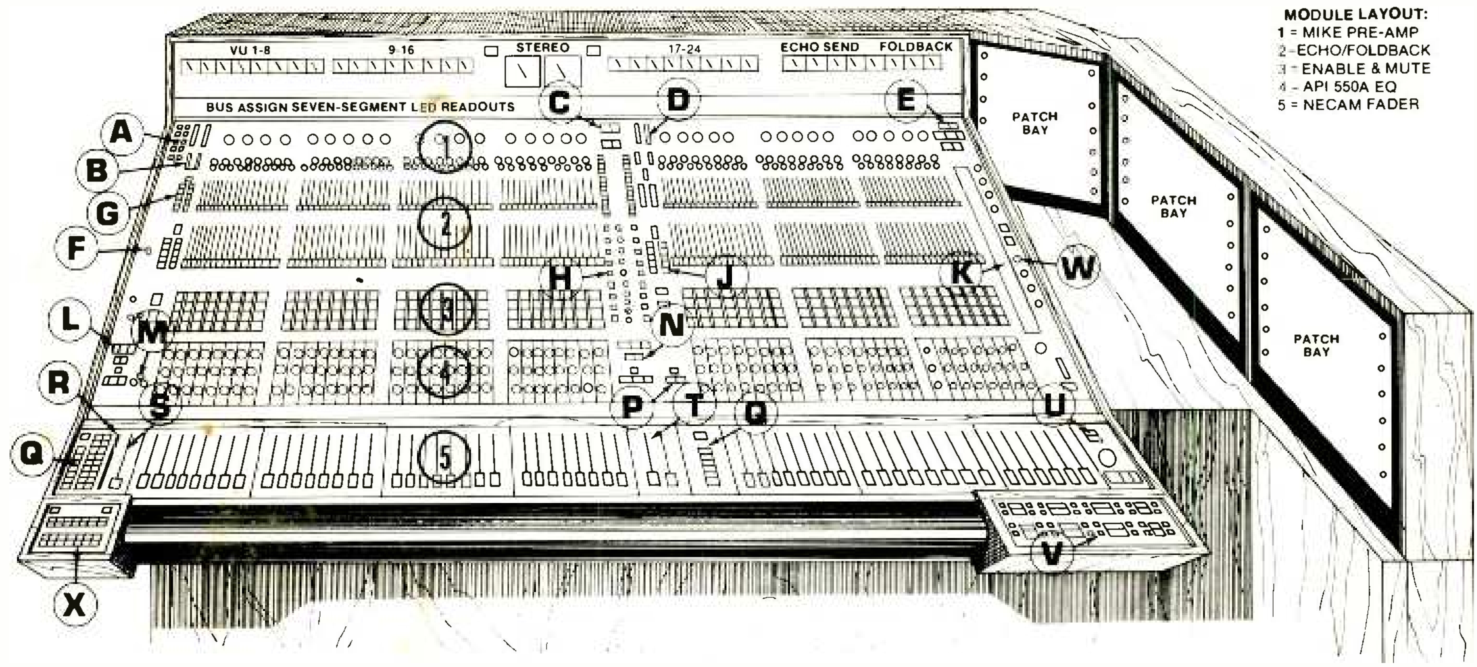
"We did most of our computations and mock-ups on [large mainframe] CAD/CAM design computers, using software that Don and I wrote ourselves. Now the company can build another console without having to run all the tests again."

How many more of these monster boards Sunset plans to build remains to be seen. In all likelihood, the studio will apply the accumulated information toward even newer devices that are destined to appear in Sunset's other studios. But there is always the possibility that the studio may decide to sell one or two consoles to parties outside the immediate Los Angeles recording scene. "Our previous console-building companies have built and sold some boards, like the one that went to the armed services," says Camarata. "But we certainly would *not* sell one to our direct competition, such as another studio in town. This board helps set us apart from the competition."

Design Concepts

And definitely it should set the studio apart. Design of the new console is unique in many respects, and reflects the latest in technology. To achieve maximum quality and headroom with minimum distortion, FETs and VCAs were eliminated from the audio path. The audio circuitry was designed using the 990 op-amp and nickel-core Jensen transformers.

"We also reduced the number of capacitors, because in some cases we found that they had more distortion



KEY TO SURFACE TOPOLOGY: A=Consolette Remote Control Logic; B=Bus Assign Programmer; C=Program Section Control; D=Same as B; E=Monitor Section Control (24-Track Playback Selector); F=Headphone Mix Select; G=Echo/Foldback Module Programmer; H=Bus Matrix Select & Submasters; J=Same as G; K=Monitor Echo Receive; L=Monitor Speaker Select & Mute; M=Clock Control; N=Same as L; P=Same as M; Q=Talkback & Room Mike; R=Monitor Selector; S=Monitor Fader; T=Stereo Bus Fader; U=Monitor to Foldback & Monitor Dim; V=Tape Machine Remotes; W=Talkback Trim & Tone Inject; X=Mute/Insert Programmer and Selector.

than we were really aware of," says Wolford. "The ultra-low-distortion circuits essentially exhibit readings that are digitally compatible: around 0.002% THD [input to output], measured at multiple frequencies — 100 Hz, 1 kHz, 10 kHz, etc. Individual stages measure what is equivalent to the residual level of our Soundtek distortion analyzer, which is one of the best devices available for such tests.

"Our goal was to design a 56-input board with Avery similar noise characteristics to a basic 32-input board," he continues. "Essentially, we've added another 24 inputs without increasing the noise. In fact, we are slightly quieter than our other Sunset boards, which are smaller and considered the quietest in the industry."

The frame also was designed on a CAD/CAM structural-design computer with rotating 3-D graphics. The computer enabled not only the stress of the console to be tested, but also the tuning within the board to eliminate any potentially audible resonating frequencies. In fact, the console now is tuned to a subsonic resonant frequency in the region of a hundredth of a Hertz.

However, the design engineers didn't restrict themselves to just high-tech circuitry and logic systems that might impress their peers, but confound the average engineer. Before anything was laid down on paper, Wolford spent many hours interviewing engineers that regularly work at Sunset. Through such impromptu market research, he was able to isolate the major operational features they preferred, and the challenges they were having with the existing boards — both Sunset's, and other stock consoles.

"We realized they wanted the console to be simple to operate, with a very clear panel layout," he offers. "That's why we used small straight-line P&G faders, rather than a lot of rotaries for the echo/foldbacks. And why we went to the central master control concept, instead of a lot of switches on the modules themselves. They also wanted some sort of indication on each function, so we color-coded all the board functions with LED indicators."

Feedback from recording engineers proved invaluable, Wolford concedes, and the intensive dialog between designers and users continued for almost two years prior to the start of actual construction. "Any time we came up with an issue — such as the three-inch fader versus a rotary control — it was always taken to the clientele for evaluation. They reviewed everything we did from the original paper sketches of the fascias to the prototypes and, finally, the finished products.

"Almost *all* of our decisions were made from an operator-ease standpoint as opposed to a purely engineering standpoint. This procedure has taken longer, but we know the finished product will be something the engineers can *use*."

Audio Path Through Console

Sunset's new board is essentially divided into two parts: one part containing 32 channels on the left-hand side; and another 24 on the right. The first 32 channel modules may function as program/input modules only, while the remaining 24 may be configured to operate as a typical "split" monitor section, complete with program/tape-replay switching, or as additional inputs. Direct outs appear on the back of each channel section, to provide up to 56 simultaneous channels on a tracking date. A total of 18 busses run through the console: 16 assignable for signal routing during recording and mixdown, and the remaining two busses dedicated to handle the main stereo outputs.

Each channel is referred to as a "section" composed of four plug-in modules: mike/line pre-amp, equalization, echo/foldback, and fader assembly. Because of the complexity of each channel section, production and design costs were reduced substantially by manufacturing identical modules for the entire board — only the faders differ from one side of the board to the other. The 32 input faders to the left contain NECAM automation, while the 24 output channels have standard Penny & Giles manual faders.

This discrete design philosophy should save time during a session, too. In the event that some aspect of a specific module malfunctions, the rest of the board is not affected. Under normal conditions, the defective module may be quickly unplugged and replaced with one that operates properly. In pressure situations, however, the engineer can simply patch around the "down" circuitry, and continue with the session.

Every echo/foldback module features an internal memory that controls all preset switching, and retains the various settings for that module. If the main AC power should be cut, the

CMOS logic memory is protected by a battery backup system that stays active for several hours, thus saving the set-up time necessary to reconfigure the various modules. Similarly, logic functions within the rest of the board are all backed up by individual battery-powered supplies dedicated to specific modules.

"The 12-volt CMOS was chosen for two reasons," says logic designer Jeff Taylor. "CMOS provides more noise immunity within the console, as a result of the higher voltage differential of the busses. CMOS also consumes very little power. By giving each module its own power supply, individual supplies can be smaller and required to furnish much less current. The largest supply is a single gel-cell battery that maintains all the relay switching within the console, like the monitor selector switching and other functions."

The circuitry within a given module is laid out so that all the digital components are isolated in one area behind a metal shield, while only pure DC is connected to the audio area. All of which ensures that digital noise doesn't contaminate the audio signals.

Signal flow through Input Channel Section.

1) Pre-Amp Module: The audio path starts with the pre-amp, which when switched to the mike position is essentially a two op-amp device. When switched to line input, the signal is injected into the device's second stage, thereby using only a single op-amp. Input and output coils of the op-amps are Jensen transformers. Both the line and mike positions feature a ±6 dB gain trim. In the mike input position, however, the gain trim is augmented with 40 dB of attenuation, selectable in 10 dB steps.

2) Equalization Module: Although it is not physically next in the external layout of the channel section, the signal now enters the EQ module — an API 550A modified with Jensen-designed 990 op-amp's, and a Jensen output coil.

3) Fader Module: Unlike other modules, which retain the same appearance regardless of their location, the fader modules differ depending on whether they are in the program or monitor half of the console. The 24 monitor channels are fitted with standard seven-inch P&G faders, while the program channels (1 thru 32) are automated with Neve NECAM servo-controlled faders. However, of the normal NECAM system, Wolford and Taylor kept only the standard P&G conductive plastic element, and the logic signaling within the fader itself. Wolford explains why the modification was made: "The typical NECAM system utilizes an internal muting circuit that actually breaks the audio path inside the fader. There is also an IC buffer amp that follows that circuit. To keep the audio path as clean as possible, we're using only one muting circuit of our own design per channel section. We developed a circuit within the echo/foldback module that receives a logic signal from the fader whenever a mute is to take place." (The mute can be controlled in several ways that will be discussed in the Echo/Foldback Section below.)

"The monitor faders aren't automated," he continues, "because NECAM is capable of controlling only 48 inputs. We decided to stop at #32 rather than automate only a portion [eight channels] of the monitor section. But we did automate the stereo output bus."

4) Echo/Foldback Module: Every channel section has four echo and four

DIGITAL CONTROL OF AUDIO FUNCTIONS

by Jeff Taylor

Approximately two years ago I was commissioned to design a digital control system for Sunset Sound's new custom console. After evaluation of the audio system concepts, I determined that two logic systems were needed: the first to control the complex audio routing within the console mainframe, including program/playback selection, monitor selection, muting functions, and solo system injection; and the second to handle switching, muting, and bus assign functions of the individual channel modules. However, both systems needed to interact for functions like solo inject and program/playback.

The first phase of the project was the development of the two "intelligent" relay cards, each of which had to offer various control inputs to allow it to be used as a universal building block. One card is a single-relay unit housing a six-pole, gas-filled relay, control circuits, current limiting lamp driver, and a delayed release circuit. The second is a dual-relay card housing two double-pole/double-throw, gas-filled relays, dual control circuits, lamp drivers and delayed release circuits. Care was taken in laying out the cards to isolate as much as possible all audio input/output and digital I/O on the card and its edge connector. Input conditioning circuits were used to allow slow rise time, and possibly noisy control lines; 12-volt CMOS logic was selected because of its lower power draw and higher noise immunity. All of the control inputs were brought out on the bottom half of a dual 22-pin edge connector; card function and behavior is a function of the strapping done on the connector in the card cage. All switching in the console handled with the relay cards is of momentary action, to allow multiple stations to command the particular relay — this also allows computer control where needed.

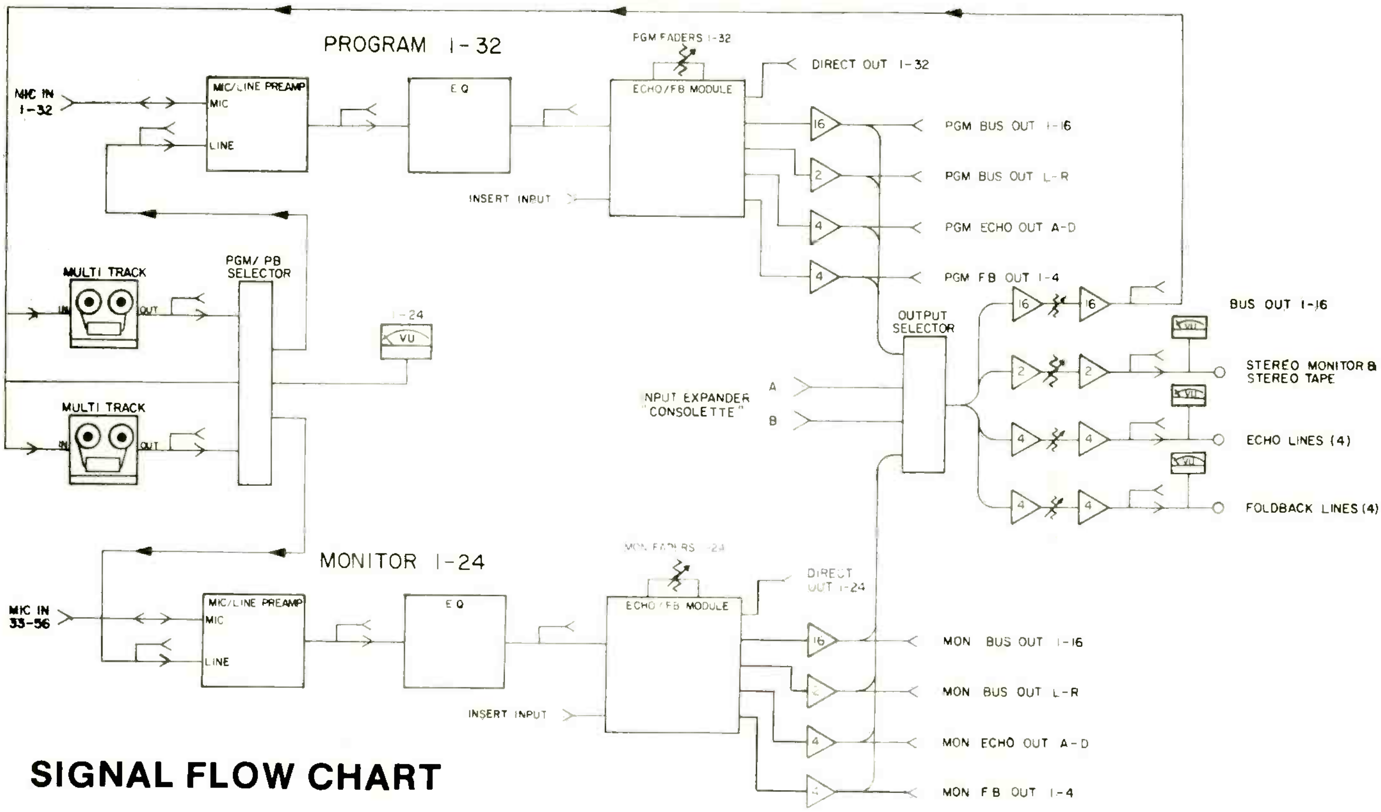
The second phase of the design was the development of the "Echo/Foldback Module" logic system, which also fell under much the same design criteria as the relay card control circuits: low power consumption, high noise immunity, and operation within close proximity of sensitive audio circuits. Because of the complexity of the switching functions required in each channel, it was decided to use a central custom keyboard instead of switches for each function on each module. At this point I looked very closely at what other console manufacturers were doing in the area of central keyboard assignment. I also talked to engineers who were using such systems, and maintenance technicians who were repairing them. The single biggest complaint from the engineers was a lack of visual indication of what the console is doing in terms of audio path routing. The largest complaint from service technicians working on today's automated consoles was the possibly catastrophic effect of a main computer going down on a session.

After hearing such comments I decided to design very intelligent channel modules. Each module has its own discrete CMOS digital processing circuitry, as well as its own memory; battery backup retains the contents of RAM in the event of a power loss. All of which results in a system that has no central computer, failure of which can cripple the entire console. Each module is connected to a 50-line, bi-directional data bus running the entire length of the console. All data on this bus comprises non-clocked, carefully shaped and timed waveforms, a requirement considered critical because of the bus being located very close to audio lines within the console mainframe. Interface points from each module were included to facilitate future computer automation of all switching functions for the entire console, including tri-state driver enable lines, and Echo/Foldback data strobe lines wired out to a card cage that will hold the computer interface.

In answer to the problem of having to work out what each module and function is currently doing, each channel module has a visual indicator for all switch functions and commands, including the status of echo and foldback sends, bus assignments, and channel muting and insert modes. Bicolor LEDs were used to save panel space.

During the packaging of the Echo/Foldback module I worked very closely with Don Wolford, who was designing the module's audio circuits. All the control logic within each module is densely packaged on a multilayer PC board, and isolated within a metal shield; no digital processing is done off this board within the module. To eliminate the risk of digital noise getting into the audio path, the audio PCBs are fed only highly filtered DC commands.

Although the system is computer-based, it functions as a stand alone system. The console has two Master Controller keyboards from which the operator can revise any switch function or bus assignment in any module. Why two Master Controllers? Primarily because the reach of a 56-input board makes it impossible for the operator to clearly see every module's status indicators from any one location. The second reason is serviceability: with two keyboards on line there is always a backup available in the event of a failure. As with all the Echo/Foldback modules, the keyboards are a modular plug-in package allowing rapid service when necessary. □□□



SIGNAL FLOW CHART

foldback sends controlled by four, three-inch P&G faders; each fader controls two circuits. The foldbacks may be run as four individual sends, or as two stereo mixes. Each stereo pair (1+2, and 3+4) has a separate panpot, so that an engineer can generate a stereo cue mix or stereo echo effects for musicians in the studio.

The echo and foldback sends may be switched to on, off, or to a mode that follows the channel muting. These conditions are controlled by two groups of six switches for the echo sends, and another two groups of six switches for the foldbacks, all switches being located on the Master Control Panel. Each mode is indicated by the appropriately colored LED: On is green; Off is no light; and red indicates Follow mode. Because the Mute indicator is red, all channels that follow the mute also are red so that an operator can see at a glance which channels are working together in the same mode. These selections are independent of the pre/post selection for each group.

A third panpot in this module may be assigned to pan in two different modes: in one setting, it pans between left and right stereo output busses, and in the other ties the panpot to the 16 assignable output busses, so the signal may be panned between any combination of odd/even numbered bus, referred to as "Left/Right" on the console switch. (How individual channels are assigned to appropriate bus outputs is covered in a later section on the Master Control Panel.)

Another slight digression is in order here, to explain why only an odd/even choice is available, as opposed to two odd or two even busses. According to Wolford, "We choose that format to minimize noise. Besides, in a pre-mix situation, most engineers prefer to put the stereo between two adjacent busses, which is naturally odd/even."

Solo and Mute Functions

Also located in the Echo/Foldback

WHY A CUSTOM CONSOLE?

According to designers Don Wolford and Jeff Taylor, many reasons warranted the design of a custom console. Sunset Sound's primary concern was to have a very high quality audio path, as evidenced in its current custom consoles. These were built using API electronics, custom panel layouts and an audio system designed for stability and purity of sound. After extensive evaluation of current stock consoles, it was determined that none offered sonic quality equal to the existing Sunset Sound custom consoles.

Several steps were taken to obtain the desired transparent sounding console:

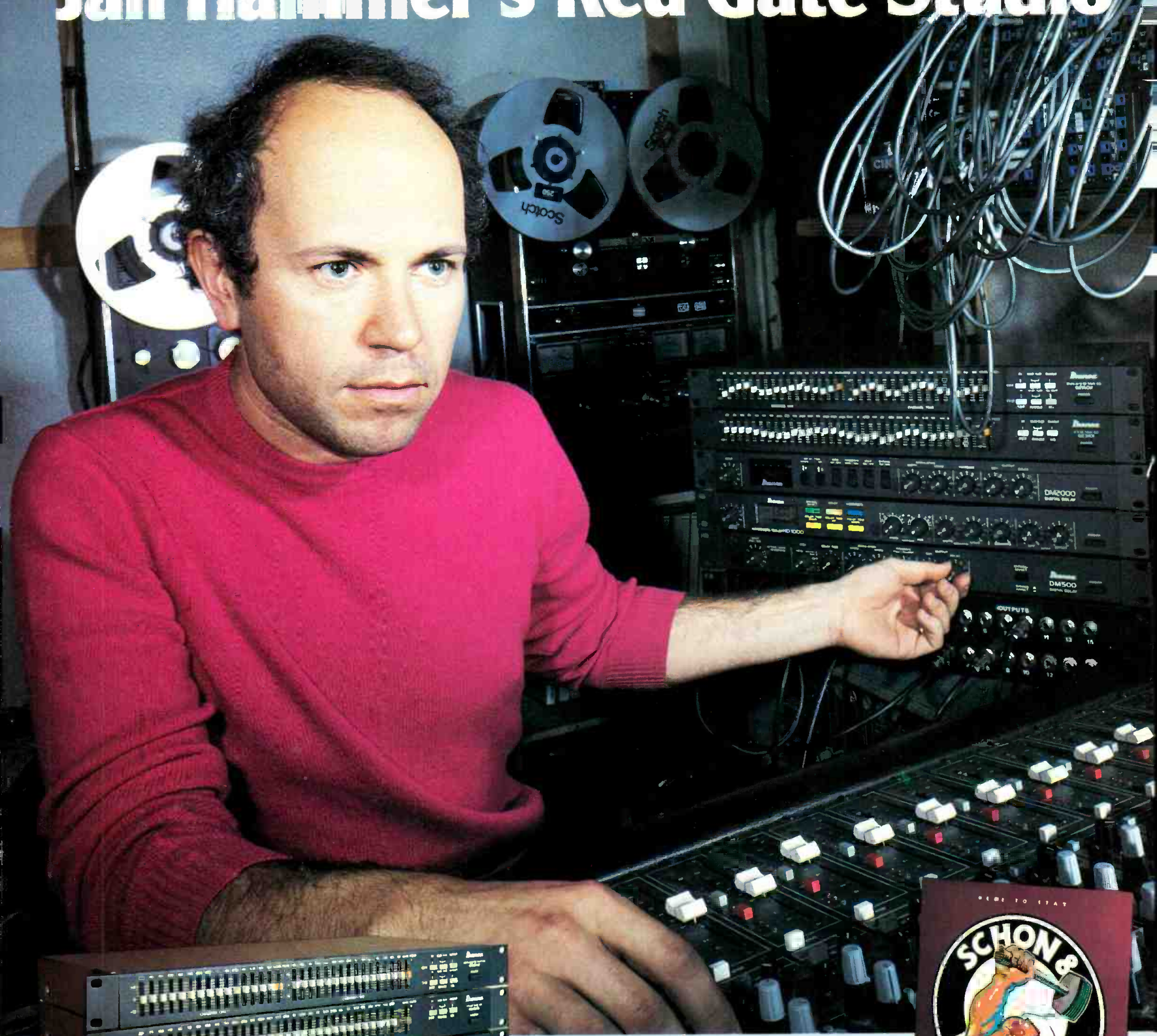
1. Development of an audio block diagram with all of the desired functions and patch points;
2. Review of gain structure, operating levels and headroom with a desire to minimize required amplifiers; and
3. Fine tuning each amplifier for maximum headroom and minimum noise using computer-aided design.

The second reason revolves around the request of Sunset's clientele to have a console with A) Separate program/and monitor sections; B) mixdown capability to two multitracks; and C) single operator compatibility. To accommodate all these functions in one console required some creative packaging. The traditional separate or "split" 24-input monitor section, when placed beside the 56 mixing faders required to handle mixdown from two 24-tracks would have made the console too long. As a solution, the console was made in two sections housing full-function inputs. Section #1 contains 32 inputs, and is called Program; the second section contains 24 inputs, and is called Monitor. Line inputs to the monitor section can select one of two 24-tracks, and feature individual Program (Bus)/Playback switches.

To enable mixdown of synchronized 24-track machines, an output selector panel was created, which selects on a bus by bus basis (1 thru 16, Left, Right, Echo A, B, C, or D, and Foldback 1, 2, 3, or 4) whether the 32 program modules and/or the 24 monitor modules are routed to the final stereo output. Two switches preset the output selector for 32-input/24-track monitoring, or 56-input stereo mixdown with full EQ, echo and foldback capability on each channel input.

The third reason came from Sunset's tradition of "easy-to-operate consoles." Status indicators and color-coded controls enable the engineer to clearly identify and operate all functions. Our fourth and final reason is reliability. The entire console was designed with gold contacts on all of the connectors, all film resistors throughout, conductive plastic faders, and the highest grade of low-distortion capacitors. □□□

Ibanez In-service: Jan Hammer's Red Gate Studio



Jan Hammer - long recognized as a pioneer of the electric keyboard - also pioneered the use of advanced signal processing techniques in his medium. Jan learned early on the value of quality signal processing in the creation of his unique musical voicings. In his Red Gate Studio Jan has chosen Ibanez digital processors and graphic equalizers. For Jan has learned what more musicians, producers, engineers and soundmen are learning everyday: Ibanez delivers!

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Module are a toggle switch for Solo, and four pushbuttons designated (from top to bottom): Set, a blue pushbutton with no label in the program section, and a Program/Playback label in the monitor half of the console; Insert; and Mute.

The *Solo* function is controlled by a three-way toggle switch that accesses the following choices: center position is off; and up triggers a red indicator, signaling that the track is being soloed. Whether or not the solo interrupts the audio passing to the tape machine is dependent upon the setting of the Summing/Mute pushbutton on the top of the output selector panel located in the center of the console. Summing solo combines individual channels to an amplifier without interrupting the signal path to the multitrack. The output, however, is a mono signal, and does not maintain any stereo perspective in the monitors. Muting solo, on the other hand, provides placement perspective in the stereo soundfield so that the stereo panning remains accurate, but interrupts the signal to the tape machine and monitors. For obvious reasons, muting solo is appropriate only during mixdown.

Finally, the toggle switch in the down position puts the channel into the Follow mode, denoted by a green LED. Follow, which works with both types of its output solo, in the summing mode the "following" channel dumps into the solo bus along with the signal from other soloed tracks. For example: if channel 18 is soloed, all the other channels on the board are muted to the monitors, unless they are in the follow mode; if #19, #20, and #21 *are* in follow, they, along with track #18, are heard in the monitors.

A channel "following" in the muting mode, however, ignored the muting command. For instance, if channel #1 in the program section is soloed, channels #2 through #32 will mute to monitors and the multitrack. However, if channel #2 also is put into the follow mode, it will ignore the muting command, and be heard with channel #1 in the monitors and on tape.

The *Set Switch*, which is actually a pushbutton, puts the channel in the mode that enables the switches on the Master Control panel. In the Off position, the logic within the module does not accept data from the master controller. When set to On, the master controller can manipulate this channel in conjunction with all other channels whose switchs are set.

Blue Pushbuttons on modules located in the program half of the board are not used in the Input mode, and therefore have no label. However, if they are plugged into a monitor channel, the blue button becomes the local Program/ Playback switch in the sync mode. (As previously mentioned, the logic in the console tells the module where it is located, and what its function is to be in that location.)

Also located in the Echo/Foldback Module of all 56 channels is an *Insert Point* that is set up post-EQ and prefader via a silent non-FET, resistive-switching circuit, and which may be accessed at the patchbay to allow a limiter, external EQ unit, or some other effect device to be inserted into the signal path.

To extend the usefulness of the insert-point feature, a series of 16 presets are installed in the lower left-hand corner of the console. An engineer can go from channel to channel setting up a series of effects, referred to as a Scene. Once the outboard gear is assigned, a simple push of any one of the 16 presets programs the switch to memorize which particular channels are affected. The operator may then set another group of channels, record those with the next pushbutton, and continue the process for a maximum of 16 scenes. Any memorized scene can be recalled instantaneously during a mix with the push of a button. For example: On the verse only, a guitar may be run through a DDL, and the piano through a flanger; these two tracks could comprise one scene stored in preset #1. On the chorus only, the same guitar may need a stereo echo effect, the piano is straight, and the horn section needs reverb: this is preset #2. Other settings can be made throughout the rest of the song for any combination of the 56 channels. During the mixdown, each of these presets may be selected at the appropriate time to recall the desired scene, and then turned off

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for the next part of the tune.

While the *Mute* function for an individual channel is self-explanatory, altogether there are four ways to access a mute on the new board, all of which are represented by the same set of LEDs. Aside from simply pushing the button, any combination of channels may be set up in a scene or group of mutes in the same way as the inserts. An additional 16 pushbutton presets reside in the same Mute/Insert Preset Keyboard area as the 16 insert presets. With one finger, an engineer can select any one of 16 preset arrangements of mutes involving all the channels on the console. A third method for muting is via the solo system. If another channel is soloed while the board is in the Muting Solo Mode, it will mute adjacent modules that are not in the follow mode.

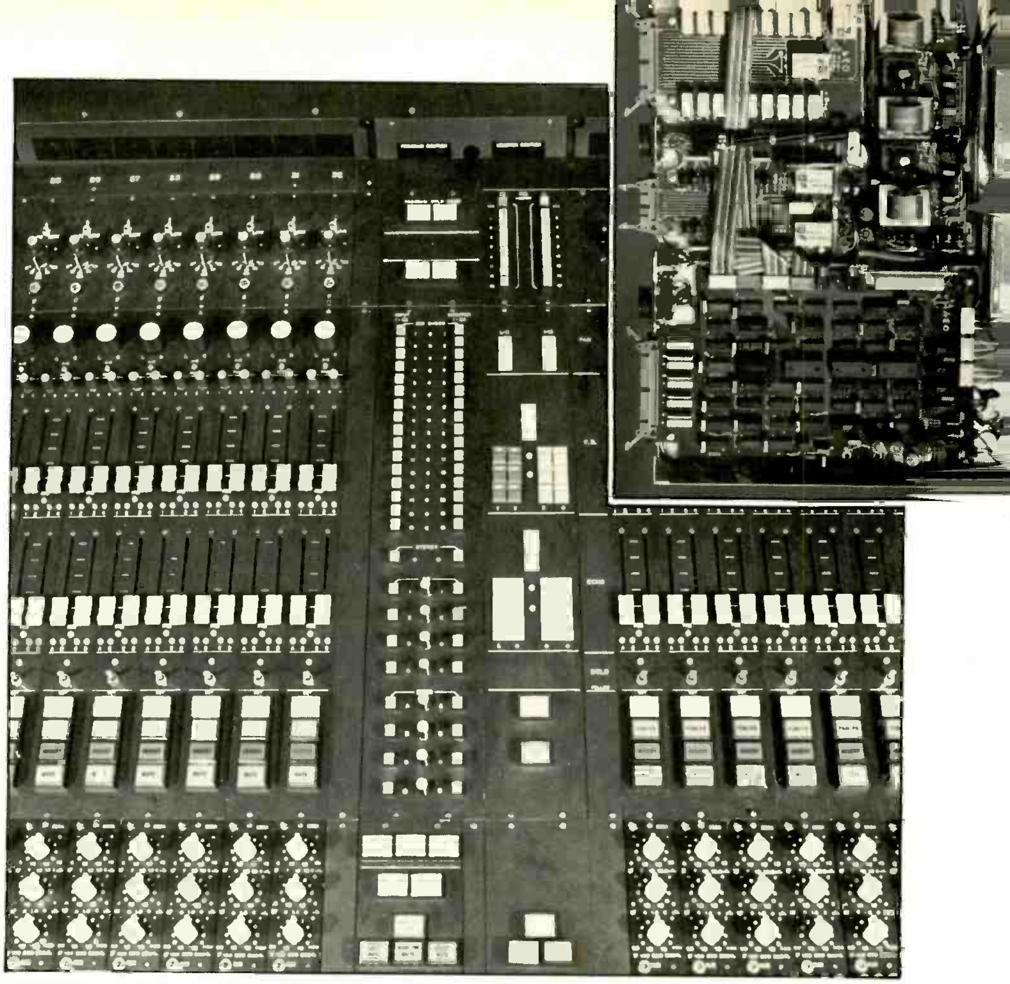
Finally, the NECAM automation fitted to the program section modules can cause a channel to mute. Because the audio path within the NECAM fader has been bypassed, the fader can now actuate the mute, which is the same type of silent switching as the inserts.

Master Control Panel

Located on the extreme left side of the console, the Master Control panel duplicates the controls also located in the center of the board; both layouts are identical. While each master panel can handle assignments for the entire console, a switch at the bottom of the panels activates one keyboard and simultaneously de-activates the other. All controls are color-coded to match the colored keys and lights on the individual modules.

"Convenience and safety are the reasons we installed *two* master controllers in the console," says Wolford. "Not only do the producer and engineer have their own control panel — so they don't have to lean across the board to change something — the duplication also acts as a safety measure. In the event that one panel goes down, the second one still operates across the entire length of the console."

The control panel handles the master selection of on, off, and follow, plus pre/post for the four echo and the four foldback send busses. At the top of the panel are located the master stereo bus on/off switch, master pan in/out switch, and master channel assign pushbuttons. Two columns of eight pushbuttons assign the output of channels to any of the 16 routing busses using the set switch at each channel, and Taylor's proprietary logic system. For example: suppose an engineer wants to route channels 1 thru 5 to bus #1. The process is as simple as turning on the set switches for those five channels to tell the computer they are ready to be programmed, and then punching the button for bus #1. To prepare for the next bus assignment, all the modules can be cleared from the master controller with one push of the Master Clear button, or individually at each module by deactivating each on-set switch.



Central control section with 36 input channels mounted to the left, and 24 to the right. Shown inset above is the interior layout of an echo/foldback module.

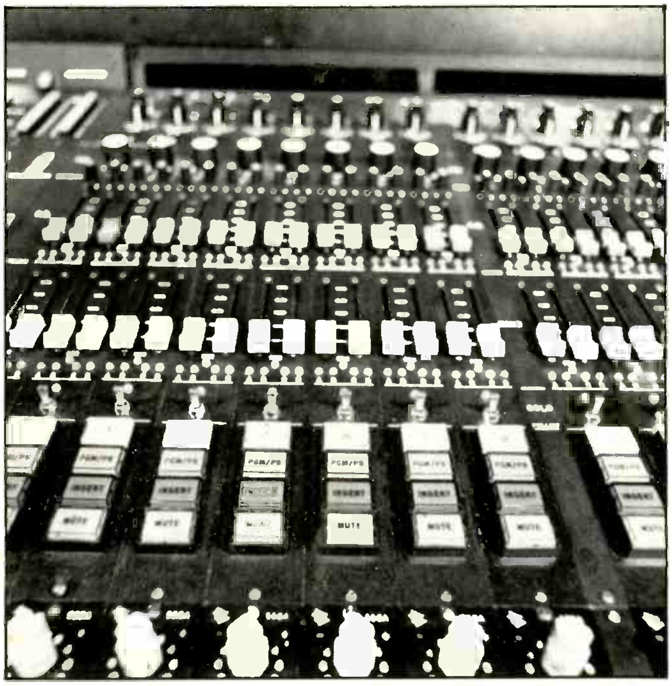
Underneath the left master control panel only, and next to the equalization modules, is the Loudspeaker Listening Section for the producer to select one of three sets of monitor speakers: control-room monitors, or either of two bookshelf sets. Located below on both panels is a set of talkback controls. The respective talkback microphones, mounted in the meter bridge directly above the master panel, can be injected into the foldback system, the studio monitor speakers, or both.

Sitting right next to fader #1 and just above the Insert/Mute Preset keyboard are 16 switches that let the control room staff listen to any one of 16 sources in the monitors. Collectively, these switches are referred to as the Speaker Monitor Selector panel, and provide a choice of 16 inputs,, including six, two-track tape machines (for replaying or monitoring sound effects, tape loops, slap echo, etc.); two auxiliary mixes (auxiliary patch points in the patchbay that carry the output of a DDL, or some other effect); four foldbacks in stereo pairs (two positions), or four mono echoes as four selections; the stereo bus set up as a final stereo remix; and the stereo bus derived from the monitor section to enable selection of just that source during tracking dates.

Output Selector Panel

The panel at left center of the board, the Output Selctor, contains two columns of pushbuttons with corresponding LEDs in the center, and assigns busses for output to the multitracks. Unlike the sophisticated logic-system used in the Master Control panel, the Output Selector is more conventional — push the button and the function happens. Bus #1 has the option of being driven from the program section, monitor section, or both, and that goes to track #1 of the multitrack. Pushbuttons on the left handle the channels from the program section on a one-to-one basis — channel #1 is controlled by pushbutton #1, and sends the audio signal to tape machine track #1. Channels 2 thru 16 are handled the same way; for channels 17 thru 24, the assignment starts over with busses 1 thru 8. Pushbuttons on the right route the audio signals from the monitor section in the same way. All channels appear at the patchbay so the engineer can swap them around to get the tracks out to the tape machines in the necessary order.

Also situated between the two columns of bus assigns are individual trim controls for each output bus. The pots for the 16 mix busses are not fitted with knobs, and are recessed. "We didn't want people to play with them and adjust them in the same way that they might with the echo/foldbacks," Wolford explains. "The trims for the tracks are 'toned' at the beginning of the session, and left alone during the actual

Close-up detail of the control surface switches, linear faders and and control elements for a group of echo/foldback and channel insert/mute/enable modules (left), and an API Model 550A equalization module (right).

tracking and mixdown."

Beneath the 16 bus assigns are 16 more pushbuttons arranged in two columns of eight buttons each, and a column of eight rotary controls down the center. Those are the master gain controls for the echo/foldback busses: four each for echo and foldback. Again, the pushbuttons in the left column refer to the program modules, and the buttons on the right handle the monitor section.

Pushbutton Mode Selection

The new Sunset console is basically 32-in by 24-out, but reconfiguration is possible almost instantaneously via the Tracking/Mixdown logic-control switch, and two columns of 16 pushbuttons that assign busses to tape machine tracks. The Tracking/Mixdown switch configures the console to the two most common modes for tracking and mixdown. In the Tracking mode, the 16 busses are assigned to the program section; the stereo output is derived from the monitor section; and the echo/foldbacks come from the monitor section. In Mixdown mode, the 16 program busses are turned off; the stereo bus is "on" for all 56 inputs; and four foldbacks and four echoes are automatically turned on for all 56 inputs.

The Tracking/Mixdown switch is intended to provide a rapid transition between the two modes. After the selection is made, the operator has the option to modify the choices with individual switches for each section on both halves of the board. Whatever is not necessary can be turned off or, in mixdown, foldback may be used as additional effects sends to bring the total to eight. Direct outs are also active to gain access to the signal path.

If, at a later date, Sunset wants to reconfigure the preset configurations for tracking and mixdown, modifications may be implemented via a pin matrix accessed from the back of the console. By simply moving any of the diode pins on a matrix, the various operating modes can be redefined for the particular session requirements.

Talkback and Accessory Panel

The panel along the extreme right hand side of the console houses several groups of function switches. "It became

. . . continued overleaf —

the area for all the extra little functions that were not used all the time," says Wolford. At the very bottom are located tape-machine remote switches. Level with the faders are the Dim Pot, Monitor Master, and Mute Relay switch, grouped together as one set. Mute Relay turns off both control room monitors at once, or mutes just the right or left channel separately. Dim Pot is a rotary potentiometer that takes the monitor level down in 2 dB steps, but is ineffective until the Monitor Dim button is pushed.

The Monitor Foldback switch routes information from the control room monitors into the headphone mix, so that the engineer and producer can monitor the same mix being sent to the studio. For two-way communication between the control room and studio during setup and recording, a separate room mike can be connected into the monitor system.

Above the talkback controls, and in-line with the fader bank, is the master Program/Playback/Sync selection. Activating the Program switch sets up the board so that the monitor channels are receiving the same signal being sent to tape machine inputs; in Playback, the multitrack outputs are routed through the monitor section. Sync simply duplicates the function of a multitrack's sync control if, like some digital machines, it does not have a discrete sync mode.

The top of the right-hand panel houses the Multitrack Selector, which determines whether the monitor section is normalled to the playback of the first multitrack machine, or the second. "This is simply another convenience switch so the engineer doesn't have to patch the multitrack machine into the monitor inputs," Wolford offers. "By hitting a switch, the operator automatically patches the second 24-track recorder into the monitor section without a single patch cord."

Comprehensive Patchbay

Even though the design of the console strives to achieve maximum flexibility through switching functions, the patchbay provides an alternative method of signal access. The bay, containing 1,032, ¼-inch sockets, is split into three sections: program, monitor, and console outputs. The program section includes patch points for mike-and line-in, EQ in/out, inserts in/out, direct outs, plus all the bus outputs from the program section. The second bay duplicates all the same functions for the 24 channels acting as monitors.

The third bay, located in the center, handles all the outputs of the output selector: multitrack in/outs, echo send/receive trunks, and foldback receive/sends — essentially, any point that is of an appropriate level for the engineer to control.

"All output channels appear at the patchbay so the engineer can swap them around to get the tracks to the tape machines," says Taylor. "Outputs are normalled so two machines may record simultaneously, and the operator doesn't have to interface a second recorder with the console."

From an operational standpoint, several reasons dictated that the patchbay be split into three parts, rather than arranged as one tall unit. Aesthetically, the lower layout is more pleasing visually and, being arced around the right side of the console, the bay is easier to access physically. In addition, a bay that is too high interfers with the monitor path, and disturbs the acoustics of a control room.

"Patch cords have been our single biggest problem in terms of liability and dependability," states Taylor. "Tracking down even a single intermittent bad connection can waste a lot of time. That's one very big reason why we put

DESIGN OF AN IMPROVED AMPLIFIER BLOCK

by Don Wolford

The criteria set for the new amplifier building block, both by myself as console designer, and Sunset Sound as user, was that it have absolute stability, meaning freedom from overshoot, ringing, or other wave-shape errors. A second concern was that the amplifier sound "transparent" when used in the studio. Additional technical goals to be achieved included maximum possible bandwidth, maximum signal-to-noise ratio, and minimum distortion. A universal package suitable for mixer, booster, or line amplifiers was desired to minimize assembly costs.

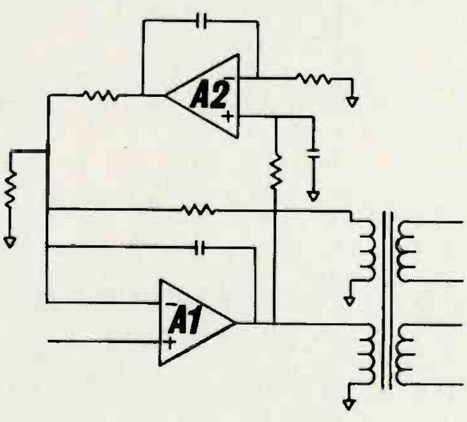
The first round of research toward these goals included a review of all component parts applied toward noise and distortion characteristics. The amplifier building block chosen was the 24-volt 990 designed by Deane Jensen, and manufactured by Hardy Company. The 990's low noise, low distortion, and high signal level handling capabilities suited our needs. An additional factor in the choice of the 990 was for its basic stability; this is best understood by examining the article entitled "Some Tips On Stabilizing Operational Amplifiers," by Deane Jensen (*R-e/p* Volume 9, No. 3, pp 42-53; June 1978).

Film resistors were selected for minimum noise, and polystyrene capacitors for amplifier stability. Non-polarized electrolytic capacitors were used for AC coupling, and metallized polyester capacitors for power supply decoupling. Bifurcated wiping, gold-plated contacts were chosen to minimize contact resistance, and to prevent distortion caused by oxidation.

Jensen transformers were chosen for minimum distortion at each of the following operating points: mike and line inputs, plus pre-amp, interstage and line amp outputs. In addition, the inclusion of an output transformer is what made possible all of the patching flexibility desired by the mixers without potential ground loop problems.

Following parts selection, a prototype amplifier was built and evaluated by the technical people at Sunset Sound. In addition to testing procedures, listening evaluations were conducted with recording engineers. During evaluation, discussions with Sunset Sound's chief technical engineer, Eric Benton, developed the concept of including the output transformer in the feedback of the 990, and removing the output capacitor.

Computer modeling and prototype evaluations followed. After several modifications to the original ideas, a circuit that met our stability specifications was optimized. This final circuit, shown below in basic form, makes use of A2 as a DC servo system, and a close-coupled transformer winding as part of the feedback. The addition of multiple feedback paths required extensive use of Deane Jensen's COMTRAN circuit analysis programs, and many hours on the HF computer in order to achieve the desired stability and waveform integrity. [For further details of COMTRAN in action, see Peter Butt's excellent two-part article in the October 1983 and February 1984 issues of *R-e/p — Ed.*] Nickel-core transformers, in contrast to steel-core, were used in conjunction with distortion-reducing feedback circuits to yield the lowest possible overall distortion in the circuit.



According to Deane Jensen, "The transformer feedback is effective from 0.2 Hz to 150 kHz. At lower frequencies the DC servo circuit provides feedback down to DC, while at higher frequencies the feedback compensation capacitor completes the feedback to 10 MHz and beyond. The 500-kHz bandwidth of the transformer, being much higher than 150 kHz, where the compensation capacitor takes over, ensures complete lack of overshoot and ringing often encountered in circuits employing transformers in the feedback of an amplifier."

The final module was packaged on a printed circuit card designed to fit into a 5¼-inch card cage holding two identical amplifiers. All connections are brought out independently for each amplifier to a dual 22-pin gold-plated edge connector. Available connections enable the amplifier to be used as a summing amplifier, or as a non-inverting amplifier with external card edge connector gain selection. □□□

Console design and assembly team: George Cross, Larry Weisbeck, Eric Benton and Mark Sackett (top row, L-to-R); Jeff Taylor and Don Wolford (sitting, L-to-R).

the patchbays at a vertical angles. Laying them down in the board would take less room, but the vertical arrangement doesn't gather dirt, and keeps the connections a lot cleaner."

Was it Worth the Time and Money?

Two questions remain, however. With such elaborate and comprehensive features as this board demonstrates, will the average walk-in engineer be able to use the Sunset console within a reasonable amount of time? "We kept the design simple and spacious for that very reason," Taylor says. "A lot of boards are more difficult to learn than this one, because the engineer has to learn how to talk to the computer to carry out any operations. This console is simply a matter of pushing a couple of buttons, and you know just about everything. There will be some questions, of course. But the seconds and mixer here will take care of those."

And, of course, the ultimate question: With so much time and money spent developing and building this analog console, isn't there a great chance that the board will become obsolete quickly within the increased movement towards digital recording?

"The boards we have now in Studio Two and Three have been used for many digital dates," points out Sunset owner Paul Camarata. "In fact, compared to the other boards in town, JVC [Cutting Center] told us that our boards are by far the cleanest of anything that's on the market now. And those consoles are about six and four years old respectively!"

"As regards our new mixing board," Camarata continues, "there's always a time when you have to stop your R&D and go into building something. On the day we stopped our research [about mid-1982], the console was technologically the most superior board anyone could have designed at the time. I don't think we could design a board that is more current with technology. This console is the cleanest sounding board possible with today's capabilities." ■■■

GETTING STARTED IN DIGITAL RECORDING AND POST-PRODUCTION

Applications of the Sony PCM-F1 for Digital Audio/Video Synchronization and Sweetening during Production of Verdi's Don Carlo for Videodisk Release

by Bob Katz

On March 20, 1983, the Metropolitan Opera performed its last telecast of the season, Verdi's *Don Carlo*, which was also videotaped with 24-track backup for later FM simulcast. However, on that day two additional audio engineers managed to squeeze into Scharff Communications' recording truck for an interesting experiment. Several things are remarkable about the supplemental recording: not only did the necessary equipment cost about $2,500, and fit easily into a small suitcase, in addition, the hardware was capable of capturing 16-bit digital audio, as well as being synchronized with videotape.

The originator of this "digital opera" experiment was New York-based audio designer and opera aficionado Vin Gizzi, who for many years has been dismayed with the quality of audio on videotape. "The audio on one-inch videotape is noticeably inferior to that of ¼-inch audio tape," he says. "The Type-C format runs at about 9 IPS, uses narrow tracks, and has an oxide optimized for *video* performance. As a result, the audio fidelity [distortion, signal-to-noise, modulation noise] suffers considerably. Even with Dolby A it doesn't compare with 15 IPS analog audio . . . plus you can imagine what happens after the three or more audio generations typically used in post-production!"

Digital At A Price

Now these are certainly depressing odds for an opera- and audiophile, Gizzi considered, but if we could somehow synchronize digital audio with video, at an affordable price . . . With visions of high-fi Butterflies (Madama species) dancing in his eyes, Gizzi asked his friend and engineer, Dave Smith, if a Sony PCM-F1 digital processor might be modified for video sync-lock. Smith not only said "yes," he also started a whole digital ball rolling at Editel Studios in New York. With his partner Kim Chinn, and help from Sony's Gus Skinas, Smith managed to accomplish the required sync-lock modification in minimal time and at very minimal cost — about $200. By then he was so enthused with the prospects of economical digital audio-for-video that he began an in-house development program to create an F1 digital editor. Since late 1983, Smith has spent all his spare time at Editel working on that project, and has made a certain amount of progress.

Until recently, as many engineers and producers discover, digital audio has been prohibitively expensive. For example, a two-channel Sony PCM-1610 costs around $25,000; you'll also need two $5,000 VCRs, and a $25,000 video editor system. Within the last couple of years, however, Japan has handed us 16-bit digital on a plastic platter: the Sony PCM-F1 and PCM-701 (F1 minus mike inputs and battery capability), and the Nakamichi DMP-100 processor — all of which cost well under $2,000, and are already being substantially discounted.

But price is not the only attractive feature of these units. Many engineers have high praise for the F1, offering that in 16-bit mode it can sound better than more expensive digital processors. Perhaps the difference can be attributed to the quality of the digital-to-analog converter in the PCM-F1, or perhaps the latter has better lowpass filtering or less phase shift. Whatever the reason, it is clear that not all digital recorders sound alike, that they are improving all the time, and that you will probably own one sooner or later.

Marc Finer, product communication manager for Sony consumer audio, explains the price differential between the 1610 and F1: "The F1 was designed to be a starter system to get consumers interested and involved in digital recording. Marketed by Sony's consumer division, it does not have the error correction capability, and therefore the dependability, of the professional 1610." [And it must be stated that the PCM-1610 offers many more features and greater operational flexibility than the F1 and its relations; there still remain many tasks that can *only* be handled by a top-of-the-line stereo digital deck — *Ed.*]

Nevertheless, an entire cottage industry is growing rapidly around professional F1 applications. It appears inevitable that the F1 will follow a long list of "semi-pro" gear that have been embraced by professionals. As Finer readily admits, "The F1 has inspired a tremendous amount of additional interest in adapting for semi-professional use." Undeniably, "semi-professional" gear can become "professional" when used by the *right* professional.

The two biggest stumbling blocks to professional use of the stock F1 are lack of editability, and video sync-lock. Videotapes recorded with an F1 cannot be edited in the normal sense, because the encoding technique was designed to fit the smaller video bandwidth available on low-cost Betamax and VHS videocassettes that would often be used to record digital audio waveforms. Aaah-so! The inscrutable Japanese construction cannot be fooled . . . or can it? Responding to the challenge, individuals and companies working in England, Germany, and America have recently managed to overcome both obstacles, which means that with proper care an audio professional can now use the inexpensive PCM-F1 as an alternative to more costly, "professional" digital recorders for most applications.

Metropolitan Opera Recording and Editing

Production of the first digital opera with pictures began at Editel in March, 1983, and was finished there in late December. Gizzi originally planned the opera recording to test the validity of Dave Smith's sync-lock modification, but word somehow got to Pioneer, who realized that a digital opera would make

— the author —
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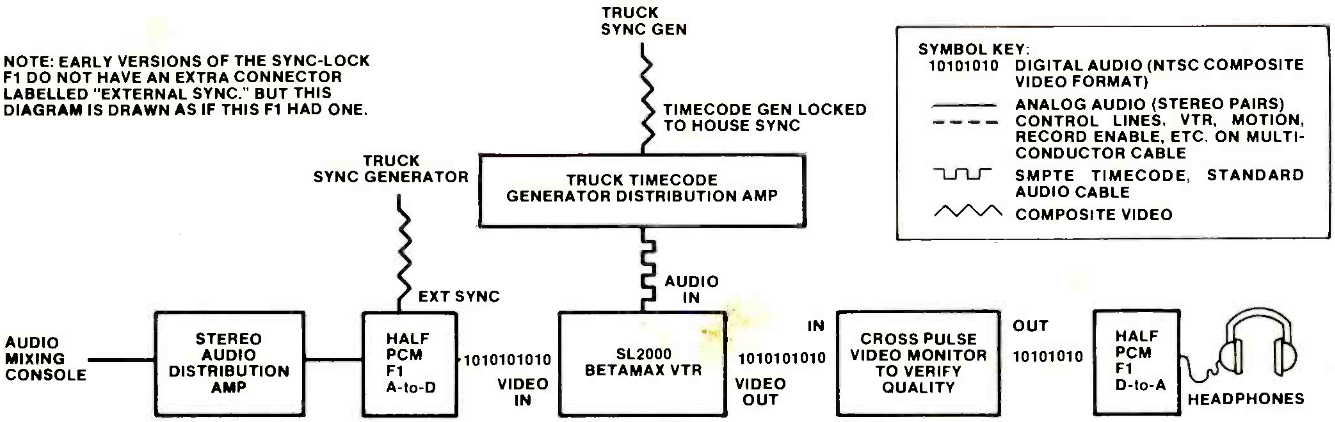


FIGURE 1: DIGITAL RECORDING OF METROPOLITAN OPERA *DON CARLO* IN REMOTE TRUCK.

a great videodisk release. Ultimately, Verdi's opera *Don Carlo* became a test of digital audio editing using a standard synchronizer, and sonic proof of the excellent quality available from 16-bit PCM-F1 recordings.

After Smith modified the F1 for sync-lock [described in an accompanying sidebar — *Ed.*], Gizzi obtained permission from the Met Opera for the digital audio experiment. "When the time came for the recording," Gizzi recalls, "I sadly found I had commitments in Europe, so I never even got to see the live performance with Placido Domingo and Mirella Freni. As soon as I knew I couldn't make it, I called my partner Francis Daniel, another musical sound enthusiast and audio engineer. Dave and Fran did the original recording on a Sony SL-2000 Betamax; hardly the hardiest medium for professional-quality digital audio.

"At that time we had no idea this would be anything but a test. Naturally, if we do it again, we'll use two ¾-inch [U-Matic] machines, one for backup, or at least two Betamaxes."

As it turned out, no audio dropouts were produced by the Betamax, which Gizzi attributes to the VCR's good condition (the video heads had recently been aligned), and the use of high-grade videotape. The resultant digital tape was also cloned to ¾-inch as soon as possible.

Here's what happened in the truck that day; Figure 1 illustrates the technical setup. Smith took reference video on a BNC cable and plugged it into the modified F1. (The sync-locked F1 strips sync information from a composite video source, and uses that signal to drive its own sync generator). SMPTE timecode was fed from an audio distribution amplifier on a cable equipped with a balanced three-pin male XLR at +4 dBv. The XLR had been adapted to a male RCA plug, thereby unbalancing the audio, and then fed directly to the SL-2000's analog audio input.

"We had experimented with recording timecode on a Betamax, and found it was tricky because of the built-in AGC [automatic gain control]," Smith says. "But we eventually found you could get recoverable code by jamming a really hot signal into the Betamax. It probably gets all squared, maybe even clipped by the machine's limiter, but somehow you get the best looking code out."

Stereo program audio was also +4 dBv balanced. Since these were isolated feeds, Smith adapted the XLRs to RCA plugs for the F1's audio inputs. "One nice thing about the PCM-F1 is it has infinite headroom at the inputs — the first thing the signal meets is the pot," Smith continues. "So as long as you don't run into hum problems, by all means unbalance the audio directly into the F1."

Before the broadcast, Met audio producer Jay David Saks sent down a 1 kHz test tone at nominal 0 VU level, the signal level being adjusted to correspond to -15 dB on the F1's meters, which are

DIGITAL RECORDING WITH PCM AUDIO PROCESSORS
An Overview of Some of the More Important Operational Criteria When Using the Sony PCM-F1

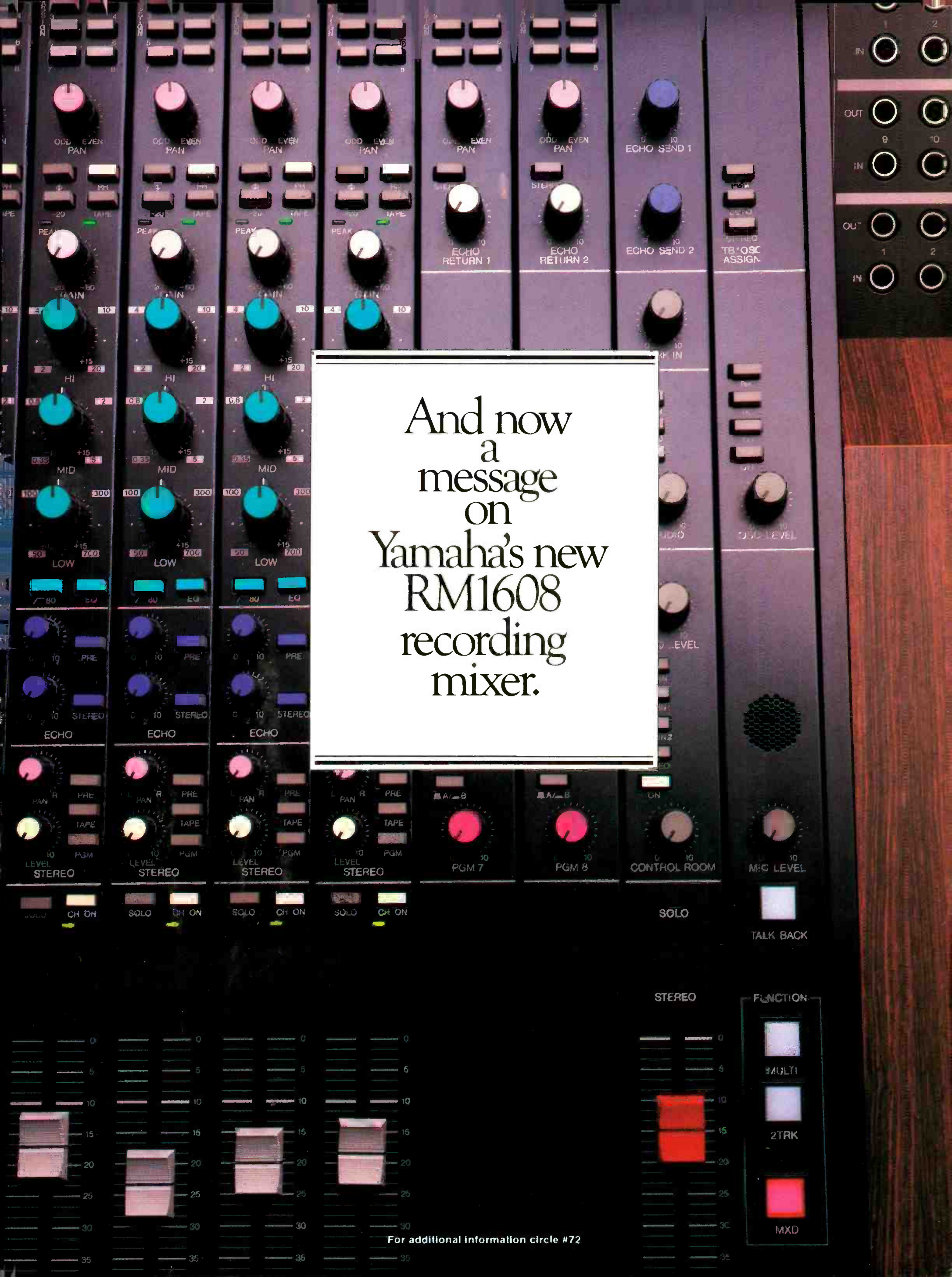
Most professionals advocate the use of two ¾-inch U-Matic videocasseette recorders, one for backup. But if portability and a two-hour recording time are important considerations, you might try using half-inch Betamax or VHS videodecks. Be sure to disable the VCR's dropout compensation and, for even more reliability, the chroma. (The "PCM" switch on the Sony SL-2000 portable Betamax performs both functions, according to Editel's Dave Smith.) One caveat: half-inch videotape is more prone to dropouts and loss of information over time. After making the original recording, it is often desirable to transfer the half-inch digital tape to a higher quality medium, such as a ¾-inch U-Matic videocassette (you can use the F1's Copy mode). Strangely, only the broadcast model U-Matic machines, such as the Sony BVU-series, will record F1 material without modification; the head switch point on an industrial U-Matic *must* be readjusted to properly record F1 material. Lastly, Dave Smith reminds users to disengage their TBCs (time base correctors) if any machines are so equipped.

The next problem to be faced is editing. Of course you *could* copy to ¼-inch analog tape with Dolby A noise reduction, and use a razor blade, but that's cheating. By renting the services of a digital editing studio, F1 material can be digitally transferred to the 1610 format for editing. Total production costs will still be economical, since a significant amount of money has already been saved by using an F1 on location. Plus there are ways to save on digital editing time by mapping edits in advance on a standard analog machine. One method makes use of timecode, since the companion Sony DAE-1100 editor requires code to be recorded on the videotape's analog tracks. During transfer to 1610, audio and timecode are recorded simultaneously onto a four-track analog recorder, which can be cut with a razor blade under less costly conditions. The resulting edit points can then be easily conformed to digital by reading the discontinuous timecode at the splice locations.

Editing PCM-F1 Encoded Videocassettes

At the time of writing, material recorded on an F1 cannot be edited directly, although various people are feverishly trying to crack the code. Several independent engineers insist that Sony will soon release an F1 editor, but company spokesmen vehemently deny that current rumor. Rumor also has it that one independent firm would reveal an F1 editor, perhaps as soon as the AES Paris Convention. At the November '83 AES in New York, Sony exhibited an industrial Betamax editing system connected to an F1, but that conflagration turned into a false alarm; unfortunately the system is only accurate to two frames (1/15th of a second), and can only edit during silence. Since most recording engineers prefer music to silence, F1 tapes must still be transferred to 1610 format for editing. Probably the best time to convert formats is during the transfer from half- to ¾-inch videotape.

Tapes can be digitally transferred to 1610 format at the editing studio via a conversion unit built by German manufacturer RTW, and distributed in the U.S. by Auditronics. The


And now a message on Yamaha's new RM1608 recording mixer.
For additional information circle #72

RM1608

SPECIFICATIONS

TOTAL HARMONIC DISTORTION (T.H.D.)
Less than 0.1% at +4dB *output, 20Hz to 20kHz (all Faders and controls at nominal)

HUM & NOISE (20Hz to 20kHz) Rs = 150 ohms
(INPUT GAIN "–60")

–128dB	Equivalent Input Noise (E.I.N.)
–95dB	residual output noise: all Faders down.
–80dB	(84dB S/N) PGM Master volume control at maximum and all CH PGM assign switches off.
–64dB	(68dB S/N) PGM Master volume control at maximum and one CH Fader at nominal level.
–73dB	(77dB S/N) STEREO Master Fader at maximum and all CH STEREO level controls at minimum level.
–64dB	(68dB S/N) STEREO Master Fader at maximum and one CH STEREO level control at nominal level.
–80dB	(70dB S/N) ECHO SEND volume at maximum and all CH ECHO volumes at minimum level.
–75dB	(65dB S/N) ECHO SEND volume at maximum and one CH ECHO volume at nominal level.

CROSSTALK
–70db at 1kHz: adjacent Input.
–70db at 1kHz: Input to Output.

MAXIMUM VOLTAGE GAIN (INPUT GAIN "–60")

PGM	74dB: MIC IN to PGM OUT.
	24dB: TAPE IN to PGM OUT.
	34dB: ECHO RETURN to PGM OUT.
	14dB: PGM SUB IN to PGM OUT.
STEREO	74dB: MIC IN to STEREO OUT.
	24dB: TAPE IN to STEREO OUT.
	34dB: ECHO RETURN to STEREO OUT.
ECHO	70dB: MIC IN to ECHO SEND.
C/R	74dB: MIC IN to C/R OUT.
	24dB: 2 TRK IN to C/R OUT.
STUDIO	74dB: MIC IN to STUDIO OUT.
	24dB: 2 TRK IN to STUDIO OUT.

CHANNEL EQUALIZATION
± 15 dB maximum
HIGH: from 2k to 20kHz PEAKING. MID: from 0.35k to 5kHz PEAKING. LOW: from 50 to 700 Hz PEAKING.

HIGH PASS FILTER – 12dB/octave cut off below 80Hz.

OSCILLATOR Switchable sine wave 100Hz, 1kHz, 10Hz

PHANTOM POWER 48V DC is applied to XLR type connector's 2 pin and 3 pin for powering condenser microphone.

DIMENSION (W x H x D) 37-1/2" x 11" x 30-1/4" (953 mm x 279.6 mm x 769 mm)

Hum and Noise are measured with a –6dB/octave filter at 12.47kHz; equivalent to a 20 kHz filter with infinite dB/octave attenuation.
*OdB is referenced to 0.775V RMS.
• Sensitivity is the lowest level that will produce an output of –10dB (245mV), or the nominal output level when the unit is set to maximum gain.
• All specifications subject to change without notice.

The specs speak for themselves. But they can't tell you how natural, logical and easy the RM1608 is to work. All the controls and switches are logically arranged to help you get the job done quickly and accurately.

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YAMAHA

For additional information circle #73

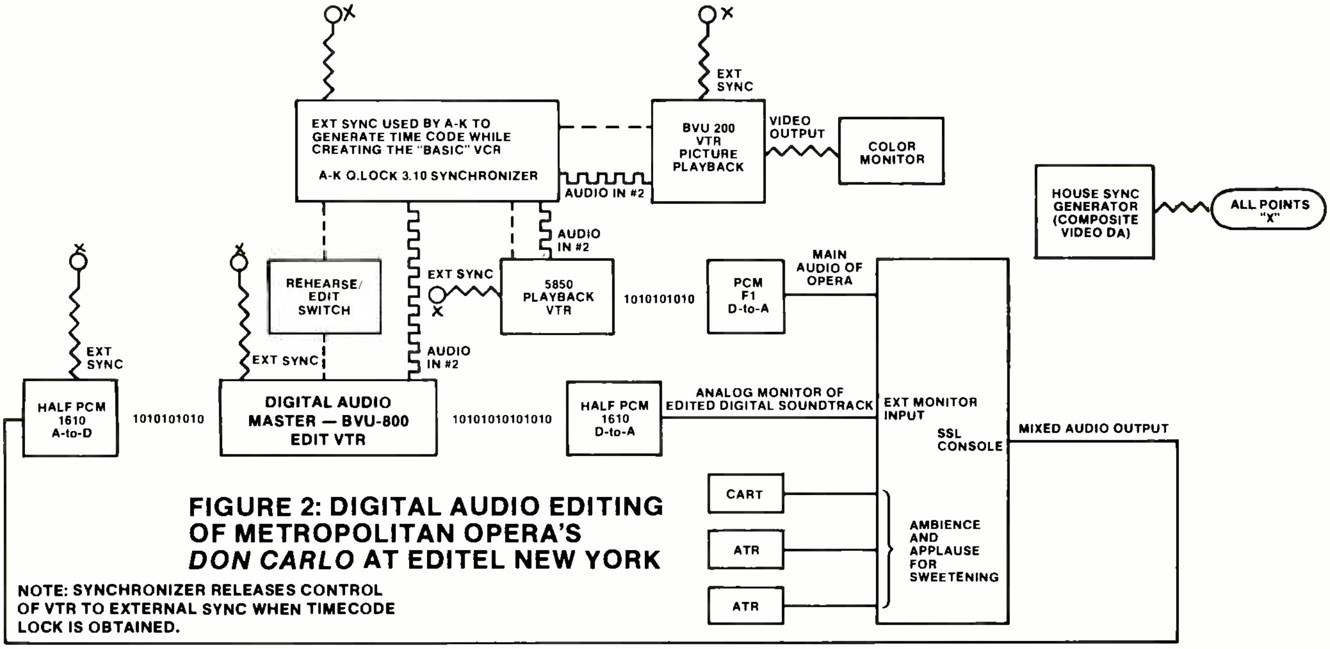

FIGURE 2: DIGITAL AUDIO EDITING OF METROPOLITAN OPERA'S *DON CARLO* AT EDITEL NEW YORK

NOTE: SYNCHRONIZER RELEASES CONTROL OF VTR TO EXTERNAL SYNC WHEN TIMECODE LOCK IS OBTAINED.

both peak reading and reflect a high-frequency pre-emphasis. (During the recording there were even a couple of peaks that went over 0 dB on the F1 metering, but no one heard distortion on subsequent playback.)

Lastly, the video output from the SL-2000 was connected to a small monochrome monitor equipped with underscan and crosspulse, so that Smith could check integrity of the digital audio signal. Smith won't leave home without such a monitor, he says; by looking at the video head switch point displayed on the screen, an engineer can tell if the video deck is within a margin of acceptability. In particular, Smith looks for a slight tearing or horizontal "slide" of the picture during the video vertical interval, which shows that all is well with the head switching.

"Then there was nothing to do except sit for two hours and enjoy the music," offers assistant engineer Francis Daniel. "Which is quite amusing, because everyone else in the [broadcast] truck had to scramble at one-hour intervals to change videotapes on the professional machines."

DIGITAL PCM RECORDING — continued . . .

RTW Analog and Digital Interface takes the F1's video output, and converts it to a 16-bit data stream that connects to the PCM-1610's digital input. According to New York engineers Ray Rayburn and Jonathan Howard, the RTW's advantages are error indication lights, and the digital conversion process. Although the box also features controllable pre-emphasis/de-emphasis, and copy-inhibit, they do not recommend using the RTW during recording because it adds noise. However, Nashville-based Rick McCollister reports having fixed a noise problem with his unit by relocating a ribbon cable.

R-e/p also has learned that a new RTW Interface device will soon be released for the Sony PCM-701 digital processor (a PCM-F1 without mike inputs and battery powering), and that Audio+Design/ Calrec is currently developing a modification kit of parts for the PCM-F1. The latter is expected to cost less than $1,000, and will provides a serial, CRC-corrected 16-bit/44.1 kHz digital output for direct connection to the 1610's digital input ports.

A direct digital F1/1610 transfer is always preferable to copying from the analog output of the F1 to the analog input of the 1610, since every conversion between analog and digital involves errors and distortions (however minimal) that should be avoided. Manufacturers and users are becoming increasingly aware of the importance of remaining in the digital domain for as long as possible. In fact, many people consider that it won't be long before most Compact Discs played in the home represent only two total conversions: the A-to-D at the original session, and the D-to-A in the listener's domestic environment. The new all-digital Neve DSP console will even eliminate accumulated conversions when working with digital multitracks, or while making equalized copies of stereo digital tapes.

After editing in 1610 format, the videotape can be transferred to Compact Disc or standard analog vinyl. When cutting a lacquer from a digital tape, a delay line must be used to provide a preview signal for the cutting lathe. Once again, try to avoid additional conversions by using a delay line that has a direct digital input. Although such devices are as scarce as hen's teeth, try to secure one for your purist recordings.

To CTC Or Not to CTC?

According to many F1 enthusiasts, tapes made for digital transfer to 1610 format should be recorded with CTC (Coincident Time Correction). Therefore, if you are planning on digitally transferring to 1610 at any time, you would be advised during the original recording to make use of the CTC® facility provided by a special unit from Audio+Design/Calrec (and which also is designed to interface -10 dBV unbalanced equipment, such as the F1, to

CTC® is a registered trademark of Audio + Design/Calrec, Inc.

Digital Editing And Post-Production

After the recording, the F1 tapes lay in storage for nine months. (Is this the natural gestation period for digital babies?) During that period, Chinn and Smith were working on their F1 editing modification in whatever spare time they could find at Editel, and Gizzi was picking up support to finish the project — that is, edit the digital tapes and create a final composite. Editel also supported the digital project by making available a state-of-the-art audio post-production room.

Finally, in December, 1983, Gizzi performed the digital audio editing; Figure 2 shows the setup used by Smith to link the Audio Kinetics Q.Lock 3.10 SMPTE synchronizer with the various video-decks. A PCM-1610 was rented for the occasion from New York Digital Recording, transfer to 1610 format being made in the analog domain.

At the beginning of the edit session, Gizzi prepared a videocassette referred to as a "basic," which eventually would become the master digital audio tape. This stage involves recording an hour's worth of house sync (usually black or color bars) onto the video track, and timecode onto the audio track of the master record VCR. For convenience, the timecode placed on the basic is made to match the code on the master videotape. After creating the basic, output of the 1610 was repatched as shown in Figure 2.

. . . continued overleaf —

During the editing process, the incoming digital audio information is recorded in sequence onto the video track of the master machine, a Sony BVU800 VCR equipped with a built-in assembly/insert video editor. A VCR's editor always enters and leaves record mode *exactly* on a video frame, thus guaranteeing a continuous, synchronous "pseudo-video" recording from end to end.

"The Audio Kinetics [Q.Lock synchronizer] is normally designed to take audio machines in and out of Record," Smith says, "but it is a simple hardware matter to convert [it] into a video editor. I took the Record Command wire from the A-K and connected it to a two-position switch, labelled 'Rehearse' and 'Normal.' In the Rehearse position the VCR goes into E-E mode [video input], so the digital audio edit could be previewed. And in the Normal position the A-K command puts the VCR into Edit. You are always listening to the output of the VCR through the 1610's D-to-A converter."

Final stages of audio sweetening and post-production at Editel's SSL-equipped dubbing room.

Gizzi's audio edits essentially conformed to the picture edits previously made at Nexus Production studios in New York. He followed an edit decision list printed by Nexus' CMX video editor, which provides in and out timecodes of each edit from the source videotapes and the edited master. Punching these numbers into the Q.Lock's keyboard, Gizzi could instruct the synchronizer to locate the VCRs to the appropriate spots. All the 1610 edits worked like a charm, he says, despite the lack of a crossfade editor. However, most of the edits were simple match cuts — that is audio edits made, for example, during applause or ambience segments — and not in the middle of music.

From Figure 2 it can be seen that some analog audio was sneaked in here and there. "But only on the transitions," Gizzi insists. "First off, we had to translate the signal to analog in order to add ambience, and match applause and other audio levels at particular edit points. But during the music, the opera's sound is 'pure 16-bit digital.'"

Such a sweetening process is quite common in video work, but usually is performed on a further audio generation. Gizzi both sweetened and edited the sound in *one* pass. Of course, instead of analog tape machines he could have used another F1 to record and replay the ambience and applause, but everyone who's heard the fine-sounding digital master tape agrees that such a process would be overkill.

Digital Audio/Video Future

As of this writing, no one knows when or if an F1 editing system will be developed. Rumors have been circulating the industry of a forthcoming commercial announcement from an unnamed source. So far, Smith and Chinn's efforts indicate that F1 material is difficult to edit without completely reconstructing the digital data. Such a digital editor appears to be a complicated device, and expensive to develop. And, sadly, an expensive editor would completely negate the PCM-F1's price advantage. On the bright side, Nashville engineer Rick McCollister now has most of the tools together to create an F1 editor with crossfade. He feels that with proper support he or someone else could come up with an economical F1 editor in less than a year's time.

As we have seen, lack of an appropriate editor has not prevented engineers from using a sync-locked F1 to obtain better quality sound for video. Gizzi edited *Don Carlo* with a rented 1610, and the sound was digital from first generation through final product (not

DIGITAL PCM RECORDING — continued . . .

the +4 dB balanced world).

CTC responds to one basic difference between the 1610 and the F1. The 1610, like most two-channel digital processors, samples both input channels *simultaneously*. However, the PCM-F1 (for cost-related reasons), alternately samples left and right channels, and sequentially feeds them to a single A-to-D converter. As a result, during the recording process the left channel lags the right by 11.34 microseconds. During the D-to-A portion of the playback process, the sampling is done in reverse order, placing an equivalent time lag in the right channel. Consequently, both channels leave the analog section of the F1 exactly in phase.

When an F1 tape is digitally transferred to 1610 format, however, the D-to-A conversion stage is bypassed, with the result that the inherent left-channel lag *remains*. CTC solves the problem by pre-delaying the right channel entering the F1's audio inputs by 11.34 microseconds using an analog all-pass network.

Is CTC necessary? Clearly without CTC, monophonic compatibility will be affected to a degree, the amount of reduced high-frequency response depending totally on the stereo miking technique. And since the Audio+Design/Calrec Propak 2 must be present in order to test for a problem, most engineers will probably use it anyway. Lack of CTC should not be audible in stereo, since the small interchannel delay is equivalent to an acoustic displacement of only 0.1 inch, which corresponds to a quarter period of a 20 kHz sinewave. (It remains to be seen whether or not the use of an analog all-pass delay network is a greater evil, in terms of additional distortion, phase error, etc., than the problem it is designed to cure; *R-e/p* would be interested in user comments.)

Cracking the Digital Code

Tapes recorded with a PCM-1610 were designed to be edited on conventional video editing systems. Editel's Dave Smith estimates that greater than 90% of typical musical edits can be performed without access to a crossfade editor (such as the Sony DAE-1100). But if cutting 1610 videotapes is easy, why is it so difficult to edit an F1 tape? "If you try to edit an F1 tape on a standard video editor, all you'll get is a mute, pop, or a click," Smith says. "One of the reasons is that at the end of every video line the F1 records a CRC word [cyclical redundancy check] for purposes of error detection. It is imperative that an editor cut the video after the proper number of cycles of CRC have occurred. We have been working on an editing system that will place the video edit at the right point."

PHIL RAMONE

ON MICROPHONES

M88RP
Printed Ribbon
Microphone

"A critical link in the recording chain is the choice of microphones. If that first stage is wrong or distorted, no amount of equalization or processing will give you back the sound that you originally intended to record.

"There are no hard rules or scientific formulas. No single mic is best for every sound. So I try to make a point to stay in touch with current microphone technology, and I'm impressed by the Fostex Printed Ribbon mics — for a number of reasons.

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"Finally, I think the Printed Ribbon technology is just as impressive in Fostex headphones. Musicians simply like to work with them. One of the best moments in the studio was when we tried the T-Series headphones and the musicians said how great it was to be able to play and really hear themselves.

"In fact, it was the Fostex T-Series headphones that prompted me to try their RP microphones.

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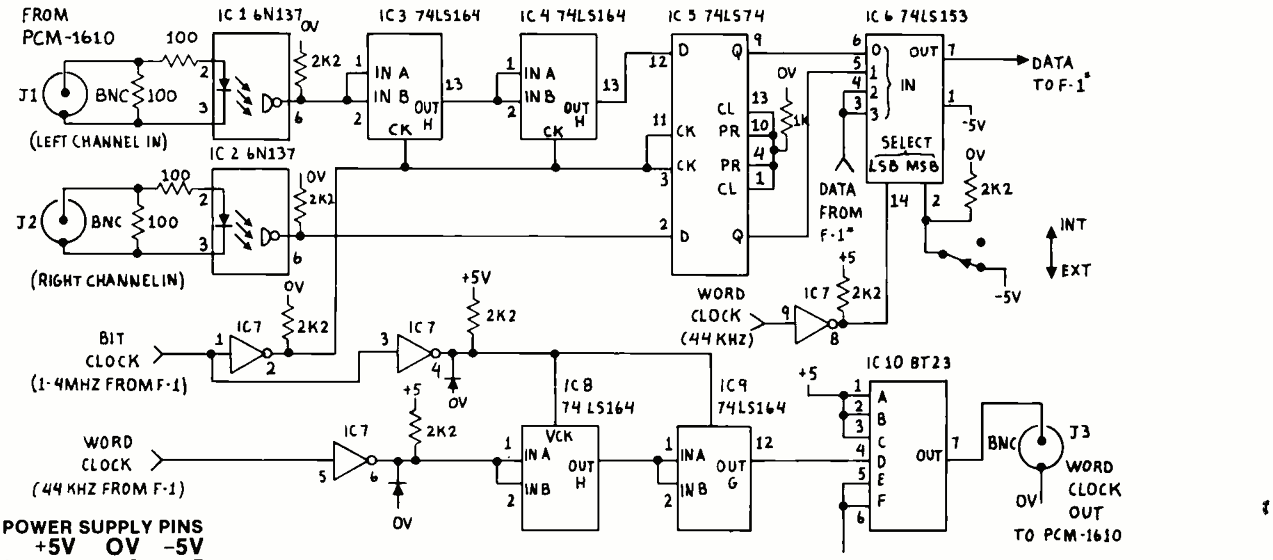
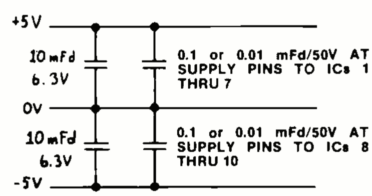


POWER SUPPLY PINS

	+5V	OV	-5V
IC1		8	5
2		8	5
3		14	7
4		14	7
5		14	7
6		16	8
7		14	7
8	14	7	
9	14	7	
10	16	8	



*Note: "Data from F-1" connects to IC506 pin #12 (playback data processor). cut the PC trace leaving this pin, and then bridge "Data to F-1" across the cut trace

Figure 3: A modification to the Sony PCM-F1 to provide direct digital-to-digital transfer of data from a PCM-1610, kindly supplied by Nashville-based engineer, Rick McCollister. To accommodate the additional components, a small circuit board can be made up and mounted inside the F1's battery compartment, or a small box built outside the unit. Power can be derived from the F1 supply. The only external change to the F1 is the cutting of one trace, as shown above; naturally, this will void the unit's warranty. Although McCollister's modification is very straightforward, he strongly recommends that a copy of the PCM-F1 service manual be obtained before opening and modifying your processor. He will be happy to send more detailed information to any interested party; contact him at the following number: (615) 356-6253.

DIGITAL PCM RECORDING — continued . . .

Editel's Smith and partner Kim Chinn are currently working on a way to cut F1 material without going through an expensive digital crossfade process, and they have had some degree of success, Smith says, "At first we thought we had the CRC problem licked; for a test we cut a 1:45 version of Michael Jackson's "Thriller," and you can't hear any edits." But classical music proved to be their nemesis: "We thought we had it made, but violins and pianissimos caused mute and click problems." Despite that setback, Smith and Chinn have not yet made plans to set fire to Lincoln Center or Carnegie Hall.

Digital Audio With Pictures: The Double System Rule

Currently, no single machine is capable of recording both digital audio *and* video on the same piece of magnetic tape. The process that needs to be used is analogous to double-system film recording, in which the ubiquitous Nagra, for example, is locked with a film camera. The "Double System Rule" is quite simple: As long as you record a signal on your audio recorder that relates to the picture, then you have created a recording that later can be *synchronized* with that picture. For example, in film work the 60 Hz pilot tone recorded on the Nagra is identical with the 60 Hz produced by the motor in the film camera. We can even use a Nagra in video work, by recording a 60 or 59.94 Hz signal derived from the video, or house sync. And, following the Double System Rule, the Nagra can be resolved later against house sync to preserve lip synchronization.

How can we make the Double System Rule work with the PCM-F1? There are many possible esoteric solutions, including recording a 59.94 Hz resolve signal on the unit's analog audio track, but the best solution involves modifying the F1. In brief, the unit's internal sync generator produces pulses exactly like the sync pulses in a standard television picture. What Smith and Chinn did was to make the F1's sync generator slave to an external source. In other words, when the recording engineer feeds composite video from the TV truck into the "sync-lock" BNC on a modified F1, the digital audio will automatically contain the proper information for subsequent video lockup. This powerful tool is not costly, either — Dave Smith says that he can modify an F1 for sync-lock for about $200; other companies will also be offering the service.

Dubbing, Timecode, and Other Niceties

We have now produced a synchronous audio recording. Even if it was made with a Betamax VCR, this digital audio recording also contains picture-lock information. How

counting the pass through an analog console). Other engineers and producers are now recording digitally, and transferring to analog videotape after the video editing, thus avoiding considerable multigeneration degradation.

On the audio-only front, McCollister has mixed over 10 albums on the F1. He swears by the Sony half-inch Betamax decks, which perform well with high-grade tape. "In well over a year I have never had a dropout," he says. "The RTW Interpolate Error indicator never lights, either."

McCollister presently cuts his albums with a pair of locked VCRs, and is currently working on a digital preview delay specifically for the F1. Other digital products up his sleeve include a means of transferring digital data from a PCM-1610 to an F1, the schematic of which is shown in Figure 3. Note that CTC (Coincident Time Correction) would *not* be needed if digital tapes were copied to 1610, and then returned to F1 format.

Other elements of the cottage industry continue to develop valuable F1 modifications and accessories, including the Meyer Sound Labs phase correction filter which corrects the phase anomolies caused by the sharp roll-off filters necessary in the A-to-D and D-to-A processors. One development on the horizon is an adapter to convert from F1 format to JVC DAS-format digital, as well as other competing systems, enabling recording to be made on a Sony,

— Editel's VCR rack —

but edited with JVC hardware, for example.

Will helical-scan videodecks continue to plague audio engineers? The use of VCRs and VTRs to record digital audio has birthed a whole set of new problems. For example, with ¾-inch VCRs it is impossible to monitor the recording until later playback, virtually requiring the use of a backup machine. [More expensive one-inch VTRs have A/B or "confidence-head" capability, but are usually cost-prohibitive for most audio applications — *Ed.*] Video heads must be changed every few hundred hours, and need to be aligned for peak performance. Editel's Dave Smith says that it is not difficult for an engineer equipped with an oscilloscope and a maintenance manual to optimize the performance of a VCR, but is yet another chore to handle prior to a session (This writer recalls the ironic claims of early digital audio enthusiasts, who said that we would be all through with alignment when everything went digital; the realists among us know that we only traded in one set of problems for another!)

In this fast-paced technological world, the future is usually here before we know it. Within about two years the industry will probably standardize on a stationary-head digital format. [Currently in the running are the Matsushita/Sony/Studer/MCI-sponsored DASH, 3M, and Mitsubishi track and encoding formats — *Ed.*] Currently available two-track stationary-head machines cost upwards of $20,000 each, and studios must plan on owning at least two decks. While expensive, stationary-head machines will be cost-competitive with the VCR-based audio processors, especially when you consider the former's ability to be razor-blade edited.

The PCM-F1, however, is considered by many to be in a class by itself; it is so economical to own and operate that smaller studios will probably prefer singing the Helical Scan Blues. Soundwise, even without an editing system, the F1 has already proved its worth in the hands of talented professionals.

■■■

DIGITAL PCM RECORDING — continued . . .

ever, unlike the professional ¾-inch decks, a consumer-type Betamax VCR cannot be synchronized with another machine during playback. Industrial Betamax machines *are* synchronizable, but most professionals advocate cloning from half- to ¾-inch U-Matic videotape. ("Clone," as opposed to "copy," is an attractive term now in vogue among the digital audio community, and is used to denote a digital-to-digital dub.) In addition, if the original digital tape was locked to a video source during recording, each of its clones will also contain the correct picture-lock information.

The above recording is *synchronous*, but is not *indexed*. Therefore it is common practice to index a recording with SMPTE timecode on the analog track, for purposes of easier synchronization and location in post-production. Note that the original timecode generator must be slaved from the house video sync generator, and not be wild or free running. And each successive digital audio clone must also contain "cloned" timecode. To dub tiecode, it should either be reshaped (squared off) or, better still, regenerated. Only a timecode regenerator that can be slaved to external video (the PCM pseudo-video leaving the playback machine) is suitable, or lip-sync will be lost. Remember that the relationship of the original timecode to house video *must* be preserved on each copy. □□□

Photography by James J March and Linda Bourassa

OBJECTIVE AND SUBJECTIVE ASSESSMENTS OF TWELVE STUDIO MICROPHONES

Stereo Testing at The University of Iowa School of Music

by Lowell Cross

The microphone is probably the only audio device whose applications have been elevated to the status of a mystique. All recording engineers have "favorite" microphones for this or that use, but it is difficult for us to articulate truly objective reasons for our various preferences. In my attempts to convey to students at the University of Iowa the basic properties of different microphones, I am often interrupted with the question, "But how does it *sound*?" Even though it cannot be answered simply or directly, this fundamental question is absolutely valid. Attempts to deal with the problem range from dispassionate scientific inquiries [1,2] to the belief that "... it's the *sound* of the microphones — and their enthusiastic acceptance — that really counts.[3]"

The present discussion does not result from electro-acoustical testing in an anechoic chamber, or from any concerns about which microphones are popular or widely accepted. This report instead presents findings obtained from a series of listening tests, made in cooperation with music students, faculty, and staff at The University of Iowa.

Evaluation Procedure

A 24-channel recording of vocal music was made in our 720-seat Clapp Recital Hall on December 14, 1983, as a class project for students in Music 25:214, Recording Techniques. The purpose of the project was to compare and evaluate the characteristics of 12 types of cardioid microphones by simultaneously recording each as a stereo pair in a reverberant environment. The human voice can reveal subtle differences among microphones as well as any possible sound source. We were fortunate to utilize the talents of one of our most promising graduate vocal majors, Carol Meyer, soprano. She was joined by Marsha Johnson, pianist, another talented graduate student who serves as staff accompaniest and as a performer in our Center for New Music. Five arias were recorded during the session, with members of the recording class serving as an audience. One aspect of the class assignment was to discuss which of the microphones came closest to approximating the "live" sound in the concert hall. The arias performed were as follows:

Roger Sessions: "Malinche's Aria" (no. 6), from *Montezuma*.

Hans Werner Henze: *"Ich gab ihr Lorbeer immergrun,"* from *Elegie fur junge Liebende*.

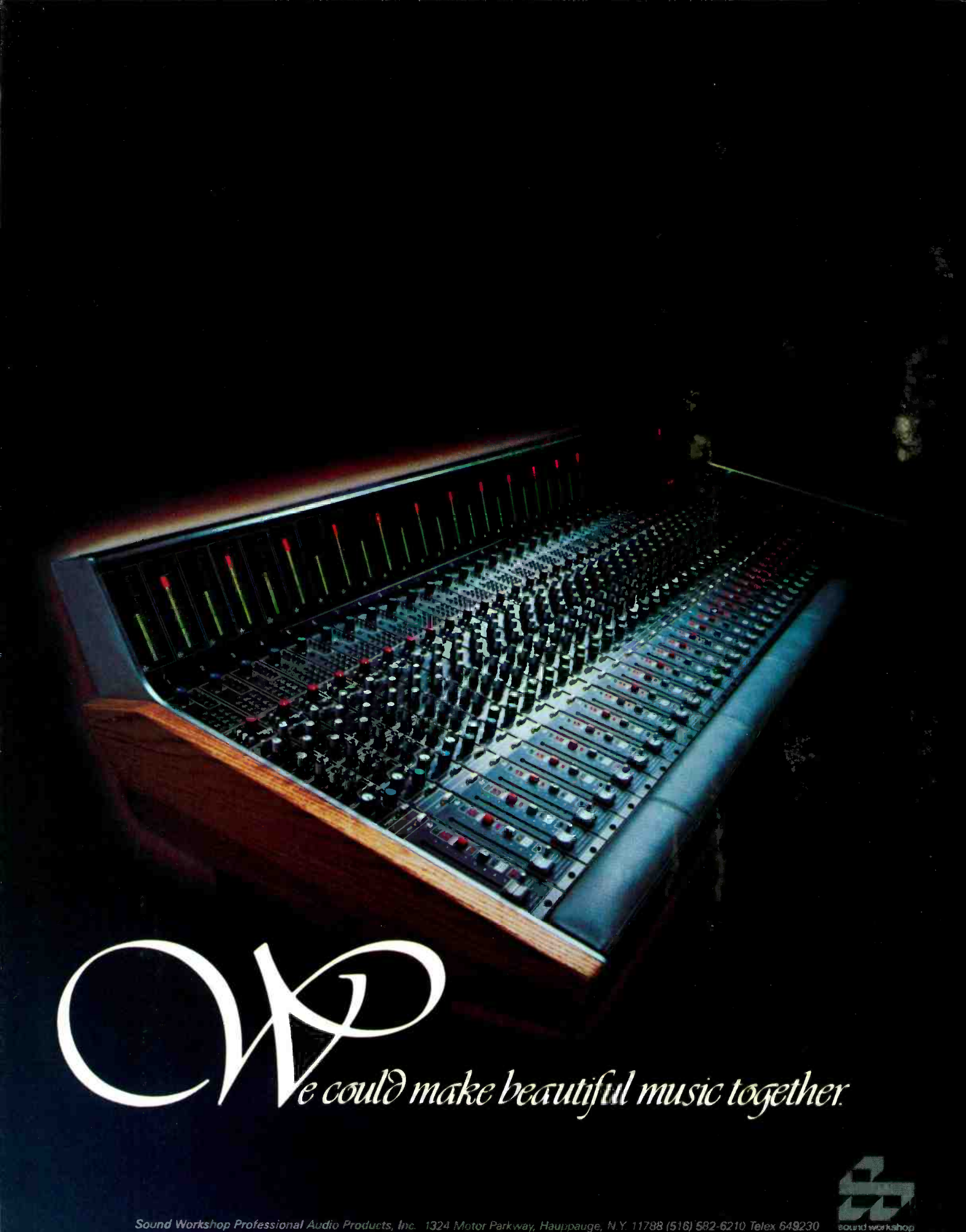
Gian Carlo Menotti: "Vola intanto," from *Amelia al Ballo*.

W.A. Mozart: "Bester Jungling (Come, my love)" from *Der Schauspieldirekton (The Impresario)*, sung in English.

George Gershwin: "Summertime," from *Porgy and Bess*.

When the session was completed, Carol, Marsha, and the class members came to our Control Studio A to evaluate the microphones while listening to each stereo pair during the playback of the 24-channel tape. The recording students and subsequent listeners have provided over 30 responses to a microphone evaluation questionnaire. As the tape was played, the microphones were announced only as pairs 1 through 12, with no intentional ordering, and with as much skipping around in the playing sequence as possible. Except for the original class demonstration, playback sessions have been limited to three or four listeners at a time, so that differences in seating positions in the studio could be minimized. Participants were invited to rate the microphones on a scale of 1 (least favored) to 10 (most favored); the results are tabulations below. (A listener could give equal ratings of "8," for example, to more than one of the microphones.)

The microphones under evaluation, all from our collection, were chosen to demonstrate a wide range of present-day and historical types. Table 1 lists



For additional information circle #77

the mikes alphabetically and numerically, with year of manufacturer, serial numbers (if available), year of manufacture, and gain requirements in the console pre-amplifiers.

None of the microphones has ever been modified in any way. All are either permanently cardioid or, if switchable among various polar patterns, were set to the cardioid position. No low-frequency rolloff or output attenuation switches were activated. Provided below are brief descriptions of each of the 12 models:

1. The AKG C452EB is a general-purpose miniature condenser microphone (18mm diameter) with a fixed cardioid pattern. It is a member of AKG's CMS family (Condenser Modular microphone System) requiring 48-volt powering[4], and offering two low-frequency rolloff characteristics in addition to the "flat" response setting of the C451 standard version. Our applications for the C452EB take into account its "bright" qualities (high-frequency emphasis); use over the cymbals in a drum set is a typical example.

2. The Electro-Voice RE15 is a dynamic moving-coil microphone with a fixed supercardioid polar pattern, providing greatest rejection at 150° off axis. It incorporates a low-frequency rolloff switch to minimize proximity effect when used close to a sound source. The RE15 and the RCA 77-DX were the only non-condenser microphones used in this evaluation. Owing to its admirable ruggedness, we use the RE15 primarily in public address and sound reinforcement applications.

3. Comparable to the AKG C452EB, the Neumann KM84 is a miniature cardioid condenser microphone (21mm case diameter; approximately 18mm capsule diameter) requiring 48-volt powering. It has no low-frequency attenuation capabilities; companion microphones offer a fixed low-frequency rolloff characteristic (KM85, cardioid; -10 dB at 50 Hz) or an omnidirectional pattern (KM83). Capsule output may be attenuated 10 dB by a switch on the microphone case. The KM84 combines excellent off-axis frequency response, good pattern integrity across the audio range, and an unobtrusive size; it is the most frequently used of the microphones in our collection for making stereo recordings of student and faculty recitals in our 200-seat Harper Hall.

4. The Neumann KM86 is a miniature, multipattern condenser microphone (omnidirectional, cardioid, and bidirectional or figure-of-eight) that utilizes two of the KM84 capsules mounted back-to-back in an almost-cylindrical mesh housing; it also incorporates a -10 dB sensitivity switch. Designed to be used at some distance from sound sources, the KM86 finds applications in recording choral, orchestral, and organ music. Engineers often use the mike for recording violin sections in studio orchestras.

Side view of multiple microphone array, showing physical orientation of cluster to vocalist and piano accompanyist used for performance recordings.

5. The Neumann SM69FET is the only stereo condenser microphone to have been used in this evaluation. It employs two, 28mm capsules of the type found formerly in the well-known U67 (tube) studio microphone of the 1960s. One of the capsules may be rotated through a 270° arc to accommodate the MS (middle-side) or XY coincident stereo techniques. This complex, versatile, and expensive microphone offeres omnidirectional, cardioid, bidirectional, and six additional intermediate polar patterns per capsule, selected by remote control on the power supply that may be located hundreds of feet from the microphone. Of the three SM69 systems in our collection, two are permanently suspended on retractable pulley mountings in recital halls (Clapp, Harper); the third is kept available for recording sessions, or for recording symphony orchestra concerts in our 2,600-seat Hancher auditorium.

6. The newest Neumann microphone, the TLM170, offers five polar patterns (omnidirectional, wide-angle cardioid, cardioid, hypercardioid, and bidirectional), MOSFET technology, and an exceptionally wide dynamic range (126 dB). It employs the same 24mm capsules found in the U89, and is notable for being Neumann's first transformerless microphone. The TLM170 has a -10 dB sensitivity switch, as well as the option of rolling off frequencies below 100 Hz. Aside from the present evaluation, we have used the microphone to record orchestral concerts, and as a solo mike (trumpet, saxophone) on jazz sessions.

7. The Neumann U47 tube microphone has achieved legendary status in the recording industry. Soon after its introduction in the late 1940s bearing the Telefunken trademark (but manufactured by Neumann), its widespread use here and abroad was instrumental in defining both the techniques and the sound qualities of many record labels: Capitol, Mercury "Living Presence," DGG, etc. With its (unfortunately) discontinued VF14 vacuum tube, 28mm capsule, and famous shape, the U47 is

TABLE 1: MICROPHONES USED DURING EVALUATION

Brand, Type, & Year Mfd.	Serial Numbers	Console Gain (approx.)
1. AKG C452EB (1978)	CK1 capsules 40059, 40098	45 dB
2. Electro-Voice RE15 (1971)	NA	60 dB
3. Neumann KM84 (1976, 1978)	28807, 36094	45 dB
4. Neumann KM86 (1971)	9121, 9122	45 dB
5. Neumann SM69FET (1971)	2094	35 dB
6. Neumann TLM170 (1983)	3/100, 4/100	40 dB
7. Neumann U47 tube (1953, 1956)	1637, 2990	30 dB
8. Neumann U87 (1971)	18610, 18612	40 dB
9. Neumann U89 (1979)	143, 144	40 dB
10. RCA 77-DX (1971)	NA	55 dB
11. Sennheiser MKH405 (1971)	4305, 4602	35 dB
12. Shure SM81 (1979)	2929, 4464	50 dB

— the author —
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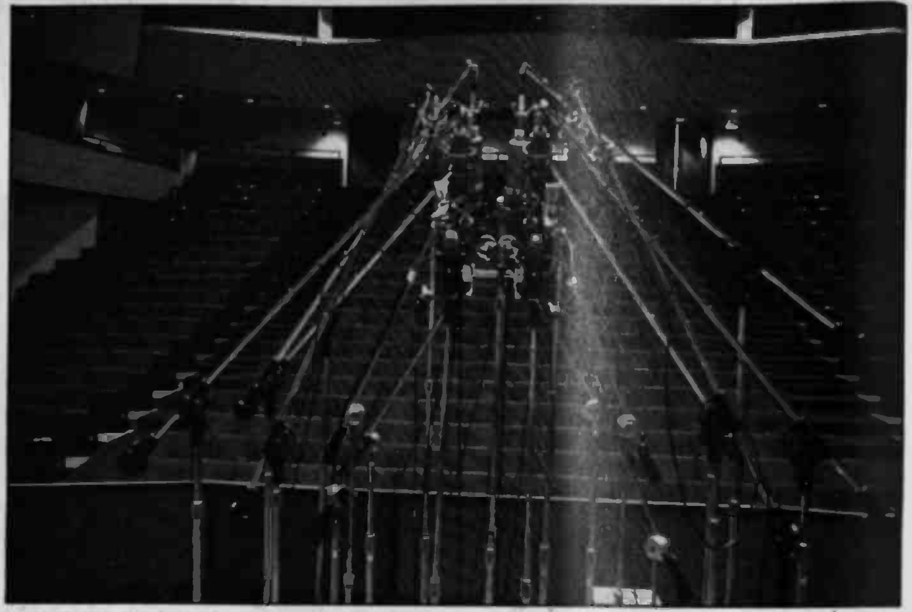
prized by collectors for its high output, robust sound, and characteristically "up-front" properties. Even though it has been used in all manner of recording situations, it is especially suited for use as a vocalist's microphone (as in the case of this evaluation). The "vintage" U47 came with a two-position pattern switch for selecting between omnidirectional and caridoid (the pattern is actually more of a supercardioid); the companion U48 permitted selection between the supercardioid and a bidirectional pattern. Today Neumann offers a field-effect transistor version, the U47FET, supercardioid only. The tube version from the 1940s and 1950s has long since become a collector's item.

8. The Neumann U87 is marketed as a studio microphone, and as such is designed to be used at relatively close distances to sound sources. It incorporates a 28mm capsule that differs in design from those used in either the U47 or the U67, its tube predecessors. Switches for three polar patterns (omnidirectional, cardioid, bidirectional), -10 dB sensitivity, and low-frequency rolloff are present; the U87 may also be powered by two 22.5-volt batteries installed internally. Our applications for the U87 usually assume a close proximity to sound sources: for vocalists (again, as in the present evaluation); instruments in a drum set; solo voices or instruments in an ensemble; and in stereo recordings of solo piano. It also is found in widespread use as an announcer's microphone in radio broadcasting.

9. The Neumann U89 is a recent arrival (1979) that looks like a "little brother" of the U87, but which costs more, features more polar patterns (the same five as the TLM170), and offers additional applications. The U89's 24mm capsule is identical to that found in the TLM170, but the former retains a transformer output. Both the U89 and the TLM170 may be used in close and distant recording situations; we frequently use the U89 to record orchestra concerts, vocal soloists, and solo instruments in jazz groups.

10. The RCA 77-DX is the only ribbon microphone used in this evaluation. Like the U47, it occupies a venerable niche in microphone history; famous performers as diverse as Bing Crosby, Elvis Presley, Groucho Marx, and the NBC Symphony Orchestra under Arturo Toscanini have relied on this ubiquitous microphone in recording and broadcasting. The natural pattern for a ribbon ("velocity") microphone is bidirectional, yet the 77-DX employs a variable shutter and an acoustical labyrinth coupled to the ribbon element to provide "polydirectional," or multiple-pattern polar response characteristics. In additional to omnidirectional (*N*, "nondirectional"), cardioid (*U*, "unidirectional"), and figure-of-eight (*B*, "bidirectional"), the 77-DX also has intermediate "limacon" patterns corresponding to wide-angle cardioid, hypercardioid, and



Vocalist's view of microphone cluster; full details of the twelve microphone pairs used during the assessment sessions are provided on the next page.

supercardioid. Two low-frequency rolloff positions (Voice 1 and Voice 2) and a "flat" setting (music) are available. The 77-DX is heavy, awkward, and fragile. Even today, it remains a favorite radio announcer's microphone, especially among AM DJs, and it is often used in studio sessions for recording brass instruments. Our applications for the 77-DX are limited to recording class demonstrations such as the present evaluation and, occasionally, in film-sound recordings or other instances when a "nostalgic" sound is desired.

11. The Sennheiser MKH405 is a general-purpose, miniature fixed-pattern cardioid microphone containing a 10 MHz oscillator; it operates on the RF modulator/demodulator principle. The MKH405 was among the earliest "transistorized" condenser microphones, introduced in the 1960s before field-effect transistors found widespread use as impedance-matching devices in condenser capsule circuitry, and utilizes A-B, or "modulation-lead" powering according to DIN 45595. The mike is characterized by high output, low internal noise, and a noticeably "bright" sound resulting from a pronounced emphasis between 2 and 10 kHz (as much as +5 dB). Originally specified by the acoustical consultant as the principal sound-reinforcement microphone for opera and stage presentations in our Hancher Auditorium, it has now been replaced there by the Neumann KM85.

12. One of the most successful of all electret condenser microphone designs, the Shure SM81 is a small, general-purpose microphone (capsule diameter approximately 20mm). Standard features include a -10 dB sensitivity switch and two low-frequency attenuation positions (6 dB per octave rolloff below 300 Hz, or 18 dB per octave cutoff below 100 Hz), in addition to a "flat" response characteristic. Applications range from public address and sound reinforcement, to broadcasting and studio recording. Like the comparable AKG C452EB, the SM81 performs well where a "bright" sound is desired.

Stereo Microphone Techniques

It is evident that all microphones used in the evaluation are cardioid (RE15 and U47, supercardioid), or adjustable to cardioid pattern. The single stereo microphone (SM69FET) was used in the MS stereo technique with a pair of matrix (sum and difference) transformers, permitting its output to be recorded as left and right channels on the 24-track tape. Since its middle element was cardioid, and the side bidirectional, it can be seen that all microphones were used as directional or pressure-gradient transducers. The pattern choice was deliberate, for the obvious reason of uniformity for all types. Furthermore, I must admit to a strong preference for cardioid microphones (*stereophony* was an important cirterion in this evaluation).

While it is true that near-perfect pressure transducers (omnidirectional microphones) are available today for specific applications[5], they must be widely spaced to achieve any kind of stereo effect or directionality. As a result, serious phasing problems occur between the stereo channels, and monaural compatibility is no longer possible. Omnidirectional pressure microphones must be carefully specified for use in either a free soundfield or a diffuse soundfield (close or distant use), since the high-frequency response can vary up to 8 dB or more at 10 kHz, depending upon the microphone's design and its distance from the sound source[6]. So, in spite of the attainments and special attractions of pressure microphones (including the absence of the low-frequency proximity effect), they cannot be recommended for stereo recordings

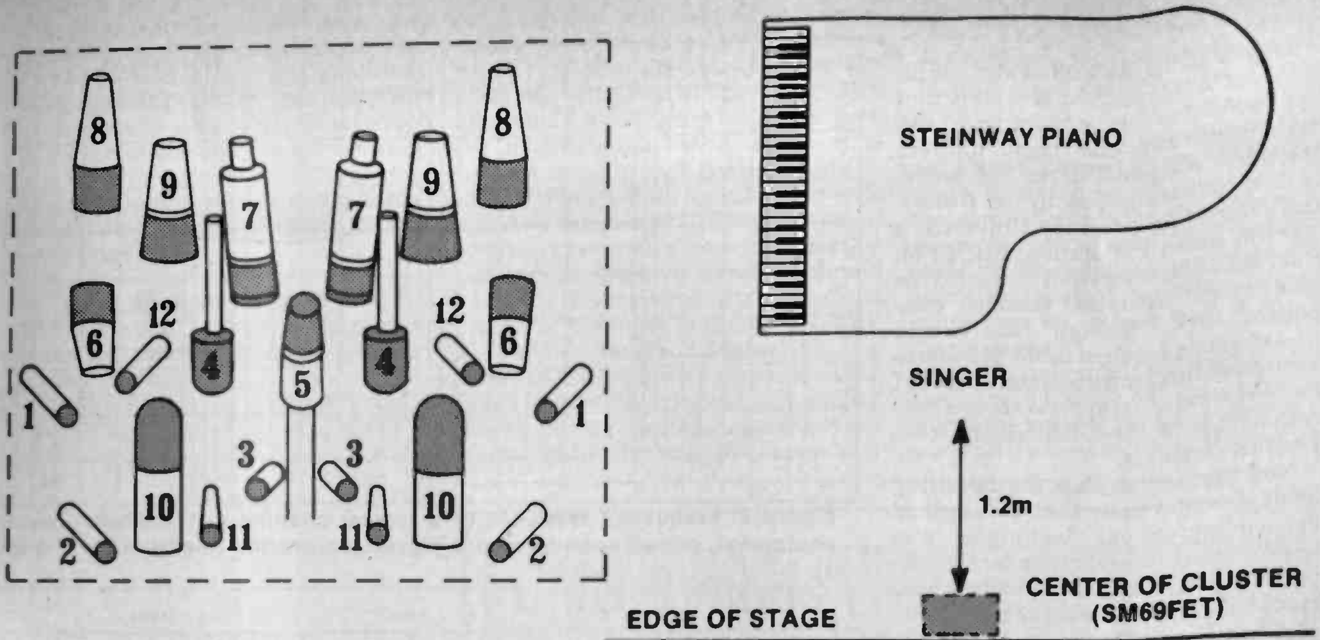


Figure 1A (left): Microphone cluster as seen from vocalist: 1 = C452EB; 2 = RE15; 3 = KM84; 4 = KM86; 5 = SM69FET; 6 = TLM170; 7 = U47; 8 = U87; 9 = U89; 10 = 77-DX; 11 = MKH405; 12 = SM81. Figure 1B (right): Floor plan of stage area during recording sessions, detailing distance between central cluster, vocalist and pianist.

employing any of the "classical" techniques.

Bidirectional microphones have been suggested for stereo recording since Alan Dower Blumlein began his pioneering work over 50 years ago. The Blumlein technique has come down to us as the use of a pair of coincident bidirectional microphones, located in the center of the stereo soundfield, and positioned so that each microphone is turned at a 45° angle away from the center line bisecting the performance space. Because of the tendency of the microphones to reproduce large amounts of the random reverberant energy developed in a typical concert hall (received by the rear lobes of the crossed microphones), the Blumlein technique demands very careful placement. Six of the microphones used in this evaluation may be switched to bidirectional patterns; indeed, the figure-of-eight is the most frequency-independent of the patterns available from any of these switchable units (KM86, SM69FET, TLM170, U87, U89, and 77-DX). But I suspect that the overwhelming majority of recording microphones in use today (for stereo or otherwise) are cardioids; cardioid stereo recording was the goal of this project. "Purists" adhering exclusively to the coincident MS, XY, or Blumleih intensity-stereo techniques may by disappointed from here on out, but coincident stereo recording simply could not be achieved in this evaluation (23 microphones in a limited space: one stereo plus 11 pairs). It is physically impossible to set up this number of microphones "coincidentally," unless some pairs are significantly more distant from the sound sources.

Moreover, I for one do not view the coincident techniques as being sacrosanct. Unless monaural compatibility is *the* paramount concern of a true stereo recording (and it invariably isn't), then why not make a judicious accommodation? One can incorporate welcome phase-difference cues — which greatly enhance depth perspective — into the stereo image by employing an acceptable spaced-microphone technique (for example, ORTF). Our own two ears perceive stereo partly as the result of phase-difference cues (recent investigations have shown that a dummy-head stereo recording, clearly a spaced-microphone technique, can work well with headphones) *and* loudspeaker reproduction: "A more detailed reproduction of the room's depth is obtained [than] the one produced by a conventional stereo microphone . . .[7]"

Regarding the MS technique, I am an adherent to this method, providing that hall acoustics are appropriate, and assuming that one must address monaural compatibility in FM broadcasting, as exemplified by our *University Concert* series on the University's 100 kW stereo FM station, KSUI-FM. But I must also report that I can hear the omnipresent monaural (middle) component back "through" the matrix transformers required in the MS stereo technique; this property disturbs me as I listen to what often apprears to be a two- rather than a three-dimensional,-two-channel recording.

Now that we can place the theorizing aside, I simply state that the cardioid spaced-microphone technique was employed in this evaluation for the simple expedient that more microphones could be placed nearer to the same "left" or "right" vantage points by slightly separating them, than any "coincidental" set of arrangements would ever allow.

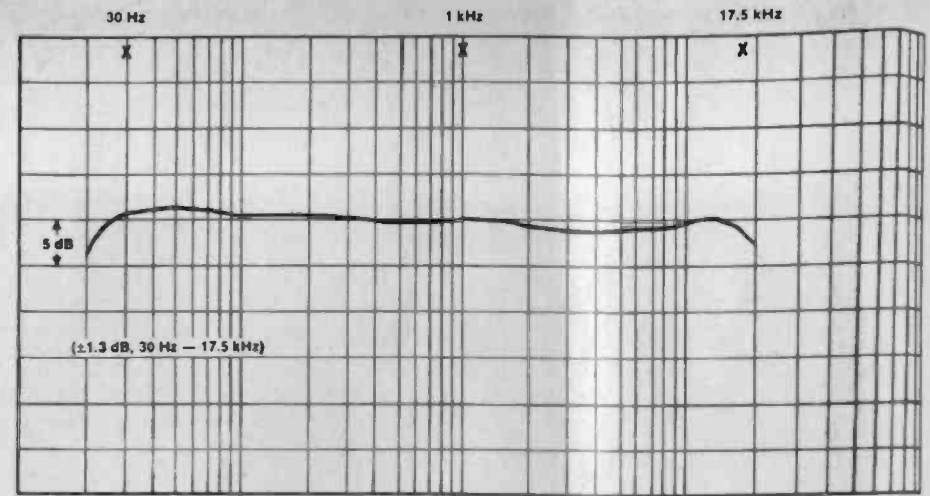


Figure 2: Frequency response of a typical channel of the Studer/Telcom equipment, overall encode/record-reproduce/decode (measured in channel 1).

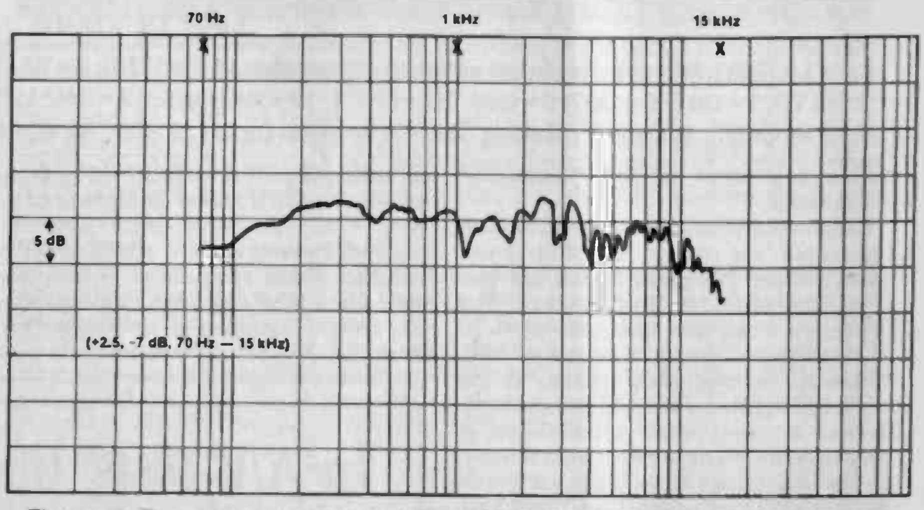


Figure 3: Frequency response of JBL 4320 monitor loudspeakers in 70 Hz anechoic chamber. (Copy of original plot.)

Concert Hall Layout and Recording Equipment

The microphone placement and positioning of the singer and Steinway 9-foot grand piano are best understood by studying the accompanying photographs and Figure 1. The SM69FET stereo microphone was placed in the center of the cluster of discrete cardioid pairs, which were all angled apart to include the soundfield as defined by the piano. The microphones were approximately seven to eight feet above the floor of the stage and, on the average, about four feet downstage from the singer. A normal center-stage concert position for the performers was chosen since the recording students were expected to consider the differences between the "live" concert arrangement, and the recording as reproduced in Control Studio A.

Clapp Recital Hall seats 720 persons, encompasses a volume of 315,000 cubic feet, and houses a 74-rank, three-manual Casavant tracker-action organ. The reverberation time for the unoccupied hall has been measured at 1.9 seconds at 1 kHz; its reverberation radius for cardioid microphones can therefore be calculated to be approximately 22 feet — our cluster of microphones was well within that distance to either the singer or the piano. To allow the quietest possible background noise level, the air-handling system for the hall was turned off for the recording session. With the turbulence from the ventilation system no longer a factor, the noise criterion (NC) for the hall is in range of 10 to 12, or 10 to 12 dB above the threshold of audibility — very quiet indeed.[8]

Equipment used during the recording and playback of the multitrack tape included: Neumann Z140 matrix (sum and difference) transformers; Neve 5315/24 mixing console; 24 channels of Telcom noise reduction in TTM frame; Studer A80VU MkIII 24-track operating at 15 IPS with Ampex 456 Grand Master tape; McIntosh MC2505 two-channel power amplifier; two JBL 4320 monitor loudspeakers (slightly modified, see below); and closed-circuit television and talkback circuits.

These brands and models of studio equipment are all relatively familiar to audio and recording engineers, with the possible exception of the Telcom noise-reduction systems. The units are still being imported into the U.S. bearing the German Telefunken trademark, but are now being manufactured by a new company, ANT Nachrichtentechnik, following the breakup and re-organization of Telefunken's original parent company, AEG. Telcom was chosen for 24-channel recording in our studios after extensive listening tests and comparisons to other noise-reduction units. Although admittedly more expensive, they offer more noise reduction than Dolby A, at least as much as dbx, and offer superb listening characteristics. Tracking and alignment problems, transient overshoot, pumping, and other undesirable side effects are minimized in these sophisticated systems. Figure 2 illustrates the frequency response characteristics of a typical channel of our 15 IPS Studer/-Telcom combination throughout the entire encode/record and reproduce/decode process. Measurements were made at reference fluxivity ("0 dB," 250 nWb/m) on the same reel of two-inch Ampex 456 tape used in the microphone evaluation recording session.

The JBL 4320 monitor loudspeakers have been slightly altered so that a greater range of high-frequency attenuation is now possible than that originally provided. The modification consisted of the insertion of simple eight-ohm L-pads between the dividing networks and the HF drivers. Figure 3 shows the frequency response of the loudspeakers after modification, measurements being made in the University's 70 Hz anechoic chamber using a one-inch calibrated B&K pressure microphone and a UREI plotter. Response readings were made 30° off of the horizontal axis at a distance of two meters (i.e., at a typical angle and distance for stereo reproduction). The UREI plotter was set for a 60-second sweep with its "slope sense" circuitry activated.[9]

The Neve 5314/24 console was used only to provide 48-volt powering, and as a set of 24 microphone pre-amplifiers during recording (all inputs adjusted for flat response — no equalization). During playback, the console served as a flat near-unity-gain switching system, allowing selection of the microphones as stereo pairs reproduced from the 24-channel tape.

Microphone Evaluations

The composite "ratings" of the microphones from the more than 30 responses were as follows:

SM69FET	8.00
TLM170	7.44
U89	7.38
U47	*7.38
U87	7.13
KM84	6.72
C452EB	6.54
KM86	6.50
SM81	6.38
RE15	5.75
77-DX	5.38
MKH405	4.27

*Tie

These numbers, of course, are subject to interpretation and even controversy. For example, the recording students present in the hall during the actual session overwhelmingly picked the SM69FET as the preferred microphone; most felt that it came closest to matching the "live" sound in the hall as the result of its reverberation characteristics. The other groups of auditioners heard only the recording while it was reproduced in our non-reverberant (I hesitate to call it "dead-end") control room. These latter groups of listeners were undoubtedly influenced by factors that they would call "presence," "warmth," etc. A particularly revealing set of composite ratings was that received from four out of five members of our vocal faculty:

U47	8.7
SM69FET	8.1
C452EB	8.0
KM84	7.5
SM81	7.2
U89	*6.6
RE15	6.6
77-DX	6.5
KM86	6.1
TLM170	5.9
MKH405	5.5
U87	5.2

*Tie

With the exception of the Sennheiser MHK405, these listeners tended to prefer microphones that emphasized high frequencies. With their operatic and concert-stage orientations, the listeners concentrated on the "brighter" vocal renditions that they thought enhanced the soprano's formant region crucial for "projection" over an accompanying instrument or ensemble. (Note that below their fifth choice of the SM81, their preferences or dislikes became much less clear-cut.)

As the author of this study, I have heard the recording more than all of the other listeners combined. Even though I admit to the subjective elements involved in an undertaking such as this, I have no hesitation in offering my own opinions. Following are my rankings of the microphones, with appropriate commentaries in each case:

TLM170 (9+). With its extremely wide dynamic range and altogether musical frequency response characteristics, this microphone displays an effortless and natural (i.e., *neutral*) quality unmatched by any of the other microphones, with the possible exception of the U89. The excellent low-frequency response of the TLM170 provides a robust, full quality in reproducing the lower octaves of the piano, while its high-frequency and transient properties seem elusively "right." The manufacturer's frequency and polar plots are revealing; the TLM170 receded slightly over the range of 3 kHz to 6 kHz (-2 dB), rising to +1 dB around 12 kHz. The microphone is not as "bright" as most common condenser units. While it does not maintain a perfect cardioid across the entire frequency range, the pair of TLM170s did offer an excellent three-dimensional stereo illusion. One student remarked that the sound was so smooth that it seemed "homogenized."

U89 (9). Only subtly different in sound from the TLM170 (whose capsule it shares), I suspect that any detectable sonic differences between my first and second choices may have very little to do with the presence or absence of transformers. The U89 appears to have very slightly more high-frequency emphasis than the TLM170, perhaps attributable to a somewhat different acoustical design of the capsule housing and screen. In all other respects these microphones are very similar indeed. An interesting observation from another student was the suggetion that this microphone was rather "cold" or "clinical" in its sound qualities.

U87 (8+). In relatively close recording situations, such as in this evaluation, the U87 is used as the manufacturer intended. So utilized, it is a neutral, accurate, and unspectacular microphone, characterized neither by gratifying "warmth" nor dominating "presence." Indeed, its unfailing accuracy in close recording is its principal virtue — but, of course, not everyone will settle for objectivity in recording.

SM69FET (8). The SM69FET received the highest composite rating from our listeners because of its ability to accurately reproduce the reverberation characteristics of the concert hall, while simultaneously projecting a clearly defined and localized stereo image. Use of the MS technique permitted the capturing of a greater amount of reverberant energy via the bidirectional capsule, than was possible from the cardioid pairs facing the sound sources. Like the stereo microphones from AKG, Schoeps, and other European manufacturers, the SM69FET is an exceptionally versatile microphone. However, it is not my personal favorite. Its rather bright qualities may be attributable to its older capsule and housing designs (after all, the capsule is the same as that of the tube U67, originally introduced about 20 years ago). I hereby propose that Georg Neumann GmbH should start building "SM89" microphones, using capsules of their own latest design as found in U89 and TLM170 microphones.

KM84 and *KM86* (8-). These microphones sound very much alike on the evaluation tape, with the KM86 offering more gratifying bass response (the KM84 incorporates a deliberate, yet slight low-frequency rolloff characteristic). They are generally not used in extremely close proximity to sound sources (neither has found much favor as a vocalist's microphone), but the KM84 can be used successfully within moderate distances over a drum set. I appreciate the accuracy and unobtrusive size of both of these microphones, especially the KM84, which also maintains an excellent cardioid pattern across the audio range. Its uniform off-axis response results in an absence of coloration when it is used in a reverberant environment (the stereo image from the kM84 pair was very well defined). But neither of these mcirophones offers much in the way of "pizazz," so they are likely to be neglected when factors other than accuracy assume paramount importance.

U47 (8–). This microphone does indeed have "character." Its robust, forward qualities, so admired by vocalists — pop, opera, and otherwise — come from an approximately 3 dB rise in the 3 to 5 kHz region (right where most of us have greatest hearing sensitivity), and another peak of about 4 dB centering around 9 kHz. I do not propose to enter the tube/transistor dialog, preferring to work with the best that both worlds have to offer.[10] The U47 tube microphone is justifiably famous; it must have been a true revelation when American recording engineers first "discovered" it in the late 1940s and early 1950s after having used all of those domestic ribbon and moving-coil varieties for so long. I enjoy bringing out our pair — power supplies, awkward cables, and all — when a recording session will benefit from their commanding presence. When using U47s, one does make a distinctive *recording*, although not necessarily an accurate one.

C452EB (6). I would appreciate our C452EBs more if their actual frequency response characteristics matched the advertised specifications. According to the on-axis (zero-degree) calibration curves supplied to us by AKG, both units are -2, +0 dB, 40 Hz to 1 kHz, and -0, +2 dB, 1 kHz to 15 kHz, instead of the ±1 dB, 40 Hz to 15 kHz claimed in the specifications. This rising high-frequency response produces the noticeably "bright" sound that was favored by our vocal faculty. My other reservation about our C452EB microphones concerns their detectable self-noise levels, made apparent (in comparison to the other condenser units) against the extreme quiet of the Studer/Telcom recording. In other respects — off-axis response, pattern integrity, and stereo imaging — these are fine, versatile microphones.

SM81 (6-). With the SM81 and companion units, Shure has demonstrated that very good microphones can be built with electret capsules. The rendition of the vocal and piano sounds was quite accurate, except for some high-frequency emphasis and a slight "edginess" in the sound. The SM81 does not appear to be as "smooth" as the other condenser microphones discussed so far, but it demonstrates admirable low-noise and stereophonic qualities.

RE15 (5-). The RE15 makes a rather good first impression. But with critical and repeated listening, its sound is revealed as being somewhat harsh and "tinny." The RE15 is deficient in low-frequency response, exhibiting a 6 dB per octave rolloff below 150 Hz (even when set for "flat" response), according to the manufacturer's specifications. The lower octaves of the piano are not properly reproduced by this microphone, and some roughness is detectable in the vocal rendition and throughout the piano range. However, these very dependable, rugged microphones play an important role in our operation; they are invariably chosen for public address or stage applications where the possibility of abuse is often present.

MKH405 (3). The low composite rating of this condenser microphone is due to its very pronounced high-frequency emphasis — many listeners find the MKH405 strident and "edgy." In all fairness, one must report that isolation transformers were required between the Sennheiser dual power supply (A-B powering) and the Neve console (wired for 48-volt powering) to prevent interference between the two powering systems. Perhaps the addition of the transformers became a contributing factor, but in all of my other encounters with the MKH405 I have invariably found its high-frequency output to be unpleasantly aggressive.

77-DX (3-). If I were a nostalgia buff, I would probably appreciate the sound of the 77-DX more. Our two RCA ribbon microphones are seldom used, and have been kept in excellent repair. Since the natural pattern of a ribbon microphone is the figure-of-eight, the 77-DX probably would have been demonstrated to better advantage with its ribbon element open on both sides (bidirectional), but this change would have negated the all-cardioid plan of the evaluation. With a limited high-frequency response and resonances in the 200 Hz and 2 to 6 kHz ranges, the 77-DX has a sound all of its own — and it is not an especially gratifying one for recording a soprano with piano accompaniment.

Conclusions

Without any prompting, or even any knowledge of which microphones she was evaluating, Carol Meyer chose the Neumann TLM170s as her clear first choice. A stereo tape of the Sessions, Henze, and Menotti Arias was duplicated from the two TLM170 tracks and submitted by Carol in a nationwide competition sponsored by the Center for Contemporary Opera in New York. We are pleased to announce that she has been selected as one of 20 finalists in the competition.

Because there are so many high quality and distinctive microphones available today, I would like to continue this process with other caridoid units, as well as with other techniques of recording: coincident XY cardioid versus ORTF versus Blumlein versus Calrec Soundfield; cardioid versus omnidirectional versus acoustical boundary (PZM™); and even tube versus transistor. I welcome the cooperation of manufacturers and importers who may be interested in loaning, or even *donating*, their microphones to The University of Iowa for future evaluations.

Notes and References

1. Boré, Gerhart; *Microphones for professional and semi-professional applications* (translated by Stephen F. Temmer); Berlin, Georg Neumann GmbH, 1978.
2. White, Philip; "Evaluation of studio microphone performance using time delay spectrometry techniques," paper presented at Audio Engineering Society Convention, 23-27 October 1982; Naerum, Denmark, Bruel & Kjaer, 1983.
3. Bartlett, Bruce; Reply to letter from Stephen F. Temmer; *Recording Engineer/Producer,* Vol. 14, No. 6 (December 1983), page 16.
4. The C452EB, KM84, KM86, TLM170, U87, U89, and SM81 microphones are powered according to DIN 45596. The SM69FET and U47 (tube) microphones each require special power supplies; the MKH405 is powered according to DIN 45595.
5. See White, *op. cit.*
6. Boré, pp 22-25.
7. Georg Neumann GmbH, Specification Sheet No. 12212 80201 for KU81 dummy head; Berlin, August 1982.
8. Reverberation time and noise measurements for Clapp Recital Hall were made by Lawrence Kierkegaard and Jeffrey Bollinger of Bolt Beranek and Newman, Inc; letter, 24 June 1974.
9. Frequency response measurements for the JBL 4320 monitor loudspeakers were made by Stephen Julstrom, The University of Iowa, 10 October 1980.
10. See: Hamm, Russell O; "Tubes vs. transistors — is there an audible difference?" *Journal of the Audio Engineering Society,* Vol. 21, No. 4 (1973), pages 267-273.

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STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

North East:

☐ **SOUNDWORKS DIGITAL AUDIO/VIDEO STUDIOS** (New York City) has expanded its company with five new divisions, one being **Soundworks Technologies** formed by engineer/co-owner **Roger Nichols**, and co-owner **Charles Benanty**. The new division will encompass studio operations, audio and video consulting, production, and all related services in the pro audio field. Nichols notes however, that Soundworks Digital will remain the center point, although he does foresee expanded technical and creative services within their studio and production realm. In accordance to this, Benanty remarks that Soundworks Technologies was formed to bring people together in hopes of benefiting themselves and the studio. "The primary aim of the division is to develop new art forms, with no format, for the creative use of technology," adds Benanty. *254 West 54th Street, New York, NY 10019.*

SOUNDWORKS' CHARLES BENANTY

☐ **KAJEM RECORDING STUDIOS** (Galdwyne, Pennsylvania) says it is among the first studios in the area to offer 24-track recording and half-inch mastering on Studer A-80 machines. Also added: an EMT 251 digital reverberation unit, and a Sony DRE 2000 digital reverb. Recent construction includes studio redesign, expansion of the isolation booths, and the construction of additional storage and maintenance areas. Acoustical redesign was overseen by **Acoustilog, Inc.** New York. *1400 Mill Creek Road, Gladwyne, PA 19035. (215) 649-3277.*

☐ **MUSIC ONE** (Syracuse, New York) is a new 16-track facility formed by a merger of **Dayson Studio** of Syracuse, and **The Music Workshop**, of Utica, NY. The studio features a customized Interface console, Scully 100 16-track with 16 channels of dbx noise reduction, and an Ampex ATR-800 two-track with half-inch heads. Monitors include Big Reds with Masterlab crossovers, JBL 4311s, Rogers, and Auratones. Reverb and delay are handled by the Lexicon 224X with LARC, Lexicon Prime Time, EMT 140 stereo plate, Eventide Harmonizer, and an Orban 111B. Outboards include Quad-Eight noise gates, Valley People Dyna-Mite gates, Orban parametric and Pultec EQ, UREI limiters, and patchable transformerless pre-amps. Musical instruments include a Mason and Hamlin grand piano, Fender Rhodes, Hammond B-3 with Leslie, Synergy digital keyboard, synthesizers by Sequential Circuits and Oberheim, DMX drum machine, and guitars by Gibson, Fender, Guild, and Yamaha. Studio manger is **Julia Scranton**. *6643 Fremont Road, E. Syracuse, NY 13057. (315) 656-8389.*

☐ **MARK RECORDING** (Clarence, New York) has added a new Lexicon PCM-42 with MEO option, a Barr plate reverb, plus a pair of JBL 4401 speakers and a Crown 150 amplifier to serve as "close-field" monitors. New microphones include Crown PZM 31-S, two AKG 451-Es with CK-1 capsules, and a matched pair of AKG "The Tube" models. *10815 Bodine Road, Clarence, NY 14031. (716) 759-2600.*

☐ **UNIQUE RECORDING** (New York City) recently updated all 14 of its digital and analog polyphonic synthesizers and sequencers to MIDI Spec 1.0. The studio has also added an AMS DMX-15/80S stereo pitch shifter/delay for rental, a second Sony DRE-2000 digital reverb for Studio B, a second Yamaha DX-7, a Sequential Circuits SX-64 MIDI sequencer, and a Roland TR909 drum machine. *701 Seventh Avenue, New York, NY 10036. (212) 921-1711.*

South East:

☐ **THE NATIONAL RECORDING STUDIO** (Baltimore, Maryland) has re-opened after some minor interior renovation to Studio A and its control room this fall. The facility now features a 3M M79 24-track, and an MCI JH-600 32/24 custom console. Also available is an Allison (Valley People) Automation system, Ursa Major 8×32 digital reverb unit, EXR Exciter, Tannoy monitors, and an array of new microphones. The studio now offers eight- and four-track recording on its newly acquired Scully 280 machine. *3016 Greenmount Avenue, Baltimore, MD 21218. (301) 467-7900.*

☐ **PHOENIX 413 RECORDING STUDIOS** (Camden, New Jersey) has installed a new Neotek Series II 32×24 console featuring four-band state variable equalization, and light-column metering on all channels. The installation and rewiring was completed by chief engineer **Mark Schultz**, and DLW Audio Consultants, Inc. DLW also designed and installed two mobile satellite racks for access to all outboard gear at any spot in the control room. Also added: a Lexicon Model 200 digital reverberation system, Lexicon Models 95 and PCM-42, 12 new compressor/limiters and expanders, and new ALS 1104 reference monitors manufactured by DLW. The entire control room was facelifted, receiving three new walls, a new floor and ceiling, new AC throughout, and a new window to the studio where closed-circuit television had been previously employed. *413 Cooper Street, Camden, NJ 08102. (609) 963-3190.*

South Central:

☐ **AUDIOVISIONS** (Louisville, Kentucky) is a new 7,000-square-foot recording complex that features two separate control rooms, both of which are tied to the main studio and an isolation booth, for simultaneous recording and mixdown sessions. The large 400-square-foot room, Control A, features a Studer A80 24-track with full dbx, Studer A80 and B67 two-track machines, and a Neotek Series III 36/24 console. A full complement of outboard gear includes products by dbx, UREI, Aphex, Lexicon, Eventide, Roland, Yamaha, White, Nakamichi, Panasonic, and Hafer. EMT 140 and Audicon plates are used for reverb. Monitoring is handled with a pair of McIntosh 2300s driving UREI 813 Time-aligned speakers. The smaller of the two control rooms, Control B, is specially designed and laid out for the varying demands of advertising and audio-visual clients. The 200-square-foot room is equipped with a Tascam 80-8 8-track with dbx, three Ampex ATR-700 two-tracks, Tascam M-50 12/8 console, Tascam and dbx outboard processors. JBL 4411s coupled to a Yamaha 2200 power amp were chosen for monitoring. Direct interconnection to all equipment in Control A is available at the patch bay. The studio itself was designed with varying acoustic properties to allow for a wide range of usage applications. Instruments include a Yamaha grand, full Yamaha drum kit, Prophet 5, Memorymoog, Crummar strings, custom Rhodes, Hohner Clavinet, Wurlitzer piano, and a variety of amps. **Glenn Meeks** undertook the acoustic design of the facility. Both control rooms are designed and built as LEDE rooms, and are currently in the process of registration. The facility is owned by **Robert Ernspiker, Tim Creed**, and **Ken Dewees**. Ernspiker, Tim Creed, and **John Schroeder** make up the engineering/production staff. Focus of operation is album production, commercial music production, and the specialized recording needs of advertising and corporate communications. *710 Distillery Commons, Lexington & Payne Sts., Louisville, KY 40206 (502) 587-6863.*

AUDIOVISIONS — New Studio Complex

☐ **PATMOS PRODUCTIONS** (Jackson, Mississippi) has added an MCI JH-24 multitrack, and a Lexicon digital reverberation unit. *Post Office Box, 16387, Jackson, MS 39236. (601) 856-2525.*

Mid West:

☐ **ACME RECORDING** (Chicago) has upgraded from 8- to 16-track with the purchase of a Tascam 85-16B one-inch deck. "I was impressed by the specs," says studio owner **Michael Rasfeld**, a veteran of 10 years' live and studio recording, "but I bought it for the sound. The frequency response looks and sounds flat, the onboard dbx is transparent, and the bottom end's tight. I imagine this machine will set a new professional standard in the workhorse 16-track format." New peripherals include a Studio Technologies Ecoplate III, Korg SDD3000 digital delay, OmniCraft noise gates, Technics five-band parametric EQ. Beyer M-88 and M-160 mike, a refurbished RCA 77-DX ribbon mike, and a number of signal processors by MXR, PAIA, and others. An arrangement with a staff member makes available a second effects rack with Ashly limiter, Deltalab 1024 DDL, Tapco Reverb, MXR Flanger/Doubler, and Korg SE400 tape echo. Onboard as staff engineer is **Todd Colburn**, and new as head technician is **Craig O'Donnell**; studio manager is **Les McReynolds**. *3821 N. Southport, Chicago, IL 60613. (312) 477-7333.*

☐ **REMINGTON ROAD STUDIO** (Schaumburg, Illinois) is the new name of Hedden West Recorders. According to studio VP/general manager **Michael Freeman**, the reason for the name change is twofold: One of the studio's founders, who'd lent his name to the facility, had left some time ago to pursue outside career opportunities, and it was mutually agreed that the studio should withold his name. Additionally, operational characteristics have improved since the studio's inception. In addition to a pair of 24-track rooms, the facility's new Tape 24 division provides real-time cassette duplication, and assists clients with disk mastering and record manufacturing. Tape 24 offers both AFGA 611 normal-bias tape, and BASF high-bias tape in a variety of lengths from C10 to C120. Dolby B and C noise reduction may be encoded at the request of the client. *1200 Remington Road, Schaumburg, IL 60195. (312) 885-1330.*

☐ **ALPHA RECORDING** (Lombard, Illinois) has added a Yamaha DX-7 digital synthesizer, a pair of Roland SDE-3000 programmable digital delays, an A/DA stereo delay unit, and an MXR Drum Computer. According to owner Bobbi Thomas, Jr. the Neotek- and MCI-equipped studio employs the only full-time, 24-track female engineer in the Chicago area, **Corinne Karpiak**, who also serves as studio manager. *515 West Harrison, Lombard, IL 60148. (312) 495-2241.*

☐ **SEAGRAPE RECORDING STUDIO** (Chicago) has updated with a 16-channel Neotek Series II console equipped with six echo returns. An MCI JH-14 16-track, with Autolocator III, and a 3M M-56 16-track have also been installed. Recently added to the studio's extensive selection of outboard gear are a Studio Technologies Ecoplate II, Lexicon Super Prime Time, and Eventide H949 Harmonizer. JBL 4430 Bi-Radial monitors are also featured. On the instrument front, the studio has added the Roland CR300 Guitar Synthesizer, according to studio manager **Mike Konopka.** *3519 W. Montrose, Chicago, IL 60618.*

☐ **TRACK RECORD STUDIOS** (Minneapolis, Minnesota) recently upgraded to 16-track with the purchase of a Tascam 85-16, 16 channels of dbx noise reduction, and remote facilities for the 16-track deck. "It's important that we meet the needs of our customers," commented owners/engineers **Norton Lawellin** and **"Red" Freeberg**, explaining that client projects had grown to where 16-tracks were necessary. The studio started as a Tascam eight-track studio, growing to 16-track in under two years. "Business has been good with the eight-track," stated Lawellin. "We're keeping the Tascam Model 38 so we can remain cost-effective in the demo field." Also recently acquired: a baby-grand piano, four Pandora limiters, Audience plate reverb, Deltalab Effectron delays, and Neumann U-87 microphones. *13912 Thomas Avenue South, Minneapolis, MN 55337. (612) 890-1075.*

Southern California:

☐ **ADVANCED MEDIA SYSTEMS** (Orange) is a new 24-track studio opened by **Daniel R. Van Patten**. The facility features a Neotek Series III-C console with eight subgroups, Studer A-80 24-track, Otari MTR-12 half-inch stereo recorder, plus Neumann, AKG, Sennheiser, E-V, Shure and PZM microphones. In-house synthesizers include a Roland system 700, System 100M, Moog Model 15, Roland Juno 60, two Roland MC-4 Microcomposers with MIDI interface and MTR-100 Digital Tape memory, Roland TR-909 Digital/Analog drum machine, Roland Vocoder, Roland CSQ-600 Digital Sequencer, and PPG Wave 2.2 with Waveterm and Expander. *833 W. Collings Ave., Orange, CA 92667 (714) 771-1410.*

THAT STUDIO — New Harrison MR4

☐ **20TH CENTURY FOX** (Los Angeles) has installed a new Harrison PP-1 console for post-production use in its new studios. The 18½-foot console has 216 inputs with three mixing positions, and will first be used for post-production work on the new movie *Rhinestone*, which stars Dolly Parton and Sylvester Stallone. **Jack Woltz**, sound director for Fox, cited three main reasons for the selection of the Harrison console: "First, we were able to get the greatest amount of capability in a console for the most reasonable price with the Harrison PP-1. Second, the delivery schedule was the best we could find. And finally, the reputation of Harrison equipment is sterling. The Disney and Goldwyn studios out here use Harrison console, and apparently are very happy with the product, which we expect to be as well." *Los Angeles, CA*

☐ **THAT STUDIO** (North Hollywood) has installed a Harrison MR4 28/24 automated console, and is upgrading its 16-track facilities to 24-track, according to studio manager **Richard Holbrook**. The studio will still offer 2-, 8-, and 16-track formats. The have also completed remodeling of the lounge and producers areas. *P.O. Box 958, North Hollywood, CA 91603. (213) 764-1421.*

☐ **IMAGE RECORDING** (Hollywood) is the new name for the facility previously known as Allen Zentz Recording, which has been sold to producer **Harry Maslin**, and previous Zentz general manager **John Van Nest**. Originally opened in 1978, the first album to be recorded in the room was Michael Jackson's *Off the Wall*, produced by Quincy Jones. As Allen Zentz Recording, the facility recorded and/or mixed numerous projects, hosting such notables as Blondie, Electric Light Orchestra, Donna Summer, and Air Supply. Since becoming Image Recording, the studio has recorded albums for Laura Branigan, Night Ranger, and Stevie Woods. Included in the sale were the original Harrison 40/32 console with Autoset automation, UREI Time-aligned monitors, various pieces of outboard gear including a 28-channel Dolby rack, EMT 140 plate, Lexicon Prime Time and Delta T digital delay systems, Eventide Harmonizer and Flangers, Valley People Kepex noise gates, and UREI limiters. New equipment purchases include an MCI JH-24 multitrack, Ampex ATR-100 half- and ¼-inch recorders, Sony PCM-F1 digital audio processor, and Yamaha NS 10M speakers. Outboard purchases include a complete line of AMS gear: a DMX15-80s stereo delay/pitch-shifter; an RMX16 digital reverb system; and two DM2-20 phaser/flangers. Also recently purchased: an ADR Panscan unit, Marshall Tape Eliminator, four Drawmer noise gates, a dbx Model rack with noise gates, de-essers and a stereo limiter/compressor, two dbx Model 160X and a 160 limiter. Instrument purchases include a Simmons drum brain and trigger system, a Roland Vocoder, and a Yamaha C7 grand piano. *1020 North Sycamore Avenue, Hollywood, CA 90038. (213) 850-1030.*

IMAGE RECORDING — Name Change

☐ **HARLEQUIN STUDIOS** (Northridge) has purchased a new Stephens 821B 24-track with autolocator, UREI 813 monitors, Yamaha NS-10 monitors, a 3M M64, 30 IPS two-track, and a new Lexicon digital reverb. In its video room, the facility has installed a complete lighting system, including lights, grid matrix, 24K dimmer packs, 12-channel lighting controller, a complete modular stage system, two JVC KY310 cameras, and a JVC/Convergence U-Matic editing system. *19347 Londelius Street, Northridge, CA 91324. (213) 993-4778.*

☐ **CANTRAX RECORDS** (Long Beach) has moved its base of operation. The new,larger studio will house equipment from Studer, Soundworkshop, Sennheiser, Electro-Voice, JBL, Bose, Tascam, Technics, Yamaha, and Valley People. Available musical instruments include Fender, Guild, Rickenbacker, Hofner, Ludwig, Zildjian, and Roland. Full signal processing and video services are also available. *2119 Fidler Avenue, Long Beach, CA 90815.*


LC-25
DC-63
DC-21
VM-41
DC-96B
XY-82
BM-73

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

Northern California:

☐ **HYDE STREET STUDIOS** (San Francisco) has opened Studio C for audio-for-video post-production. The room features a BTX Shadow, Otari MTR-90, API console, Sony VCRs, and custom computer control. The studio has also expanded its collection of rare signal processors with the addition of a Fairchild tube limiter, and several Pultec tube equalizers. Also added: another Otari MRT-90 16-track wired for 24-track recording. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

☐ **STUDIO C** (Stockton) has added a new Otari MTR-10 two-track tape machine. *2220 Broadridge Way, Stockton, CA 95209. (209) 477-5130.*

☐ **PATCHBAY STUDIOS** (San Rafael) plans to install Meyer Sound Labs 833 control room monitors. "A recent demo of these remarkable monitors convinced us that we will have one of the finest mixing rooms in Northern California once they're up and running," a studio spokesman said. *2111 Francisco Boulevard, San Rafael, CA 94901.*

AAV — The Roadie Mobile

☐ **BEGGAR'S BANQUET RECORDING STUDIOS** (Santa Rosa) has added a Lexicon 200 digital reverb unit to its list of outboards. *540B E. Todd Road, Santa Rosa, CA 95407. (707) 585-1325.*

Australia:

☐ **AAV-AUSTRALIA** (Melbourne) has taken delivery of a Sony PCM-3324 digital multitrack, described as the first such machine to be put into service in the Southern Hemisphere. Along with the 3324 the studio complex has added a PCM-1610 digital stereo mastering system, and DAE-1100 editing processor. Also available the Sony PCM-F1 system. Available for mobile recording dates, The Roadie, unveiled late last year, features a Sound Workshop Series 40 32/24 console with high-resolution LED metering, two 24-tracks with Dolby, one two-track, plus Tannoy Little Red and Auratone Super Cube monitors powered by Crown DC300A power amps. *180 Bank Street, South Melbourne, P.O. Box 296, Victoria 3205, Australia. (03) 699-1844.*

— *AUDIO/VIDEO UPDATE* —

Eastern Activity:

☐ **SMALL BIGGIE BURNS** (New York City) is the name of a new production company that specializes in music-video programming for television, motion pictures, cable TV and home video. Principal partners are **Robert Small**, **Alan Hecht**, and **James Burns**. The company's first two music-video projects were Charley Pride's "Every Heart Should Have One," and K.C.'s "Give It Up." The trio first collaborated as creative consultants for *One Night Stand: A Keyboard Event*, starring Eubie Blake and the Jazz All Stars; the program received a Grammy nomination for Best Video of the Year in 1981. Other credits include a music video for Calamity Jane's "I've Just Seen a Face," and art direction for the Stray Cats' hit "Rock This Town," Rachel Sweet's "Be My Baby/And Then He Kissed Me," and for the Producer's MTV video clip "What's He Got." *36 West 62nd Street, New York, NY. (212) 245-7385.*

Western Activity:

☐ **CINEMAN** (Hollywood) is a new company that specializes in the production of music videos. The new team consists of director and video artist **Derek Chang** and director **Steve Lewis**, formerly of *Solid Gold*, producer **Tony Basile**, and associate producer **Diane Beam**. Chang's work includes videos for A&M's Horizontal Brian ("Practicing First Aid") and Vivabeat ("The House is Burning"). "My videos," he says, "are very personal. I believe in making a statement, but at the same time being subtle about it, and being entertaining." Basile has been responsible for over 300 television commercials, including such clients as Datsun, Toyota, McDonalds, General Tire, and Disneyland. Over the past three years he's produced several music videos, including The Motels' "Remember the Nights," Diana Ross' "Muscles," J. Geils Band's "I Do," Juice Newton's "Angel in the Morning," and "Queen of Hearts," Dave Edmond's "Information," and Tom Jones' "The World of Tom Jones." *2242 Cahuenga Boulevard, Hollywood, CA 90068. (213) 461-0360.*

☐ **EFX SYSTEMS** (Burbank, California) announces that **Debra A. Knetz** is the new studio manager for the post-production and audio sweetening facility. *919 N. Victory Boulevard, Burbank, CA 91502. (213) 843-4762.*

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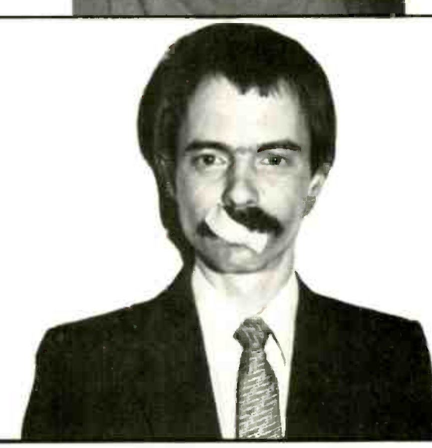

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Photography by Harry Butler

HUMMINGBIRD STUDIOS, NASHVILLE

An Eighth-Floor Jingle and Commercials Production Facility with an Interesting Set of Sound-Isolation Problems

by Sam Borgerson

The folks at Hummingbird Productions in Nashville have made an astounding discovery: Recording studios don't make music; People do.

At first glance, such a statement might seem obvious to the point of silliness. But, when you stop to think about it, how many studios take all the human factors fully into account from the earliest design stages? Furthermore, how many studios relying primarily on commercial music clients are designed, from the outset, for the unique demands of jingle production?

Not many, according to the people at Hummingbird.

Of course, staff producers at Hummingbird would readily admit that their demands go beyond those of most jingle producers, primarily because Hummingbird is a very people-oriented company. You can sense this by listening to a Hummingbird demo reel: there's a lively, bouncy, uplifting feel in here that cannot be electronically synthesized. So when Hummingbird set out to build its own studio, three human factors were paramount in determining the design: comfort, communication, and inspiration.

But first, some historical background. Hummingbird Productions was started by Bob Farnsworth and his wife Merrill in 1976. Farnsworth had come to Nashville with a cousin in 1975 and, as a folk-rock duo, they were signed by ABC. They released one album which, according to Farnsworth, "started off slow and tapered off from there." Dropped by the label, Farnsworth was on the verge of returning to Greenville, South Carolina, to join in his father's insurance business when he decided to take a stab at writing jingles.

This time, something clicked. Farnsworth found he had an aptitude for the art, but his success was undermined by poor treatment from the jingle firm employing him. So, on $30 of borrowed cash, the Farnsworths started their own commercial music business. On another $500 of borrowed money, Farnsworth made the first Hummingbird demo. Fortunately, it sold; otherwise Hummingbird would have gone belly up. He wasn't going to go any deeper in debt.

The company grew rapidly, pulling in dozens of local and regional clients. The big break came when they submitted a demo to the "Big Mac" of all jingle clients. Although somebody else got that national campaign, McDonald's agency (Needham, Harper, and Steers) sent Hummingbird some other business. Later, Hummingbird was awarded the Ronald McDonald campaign, aired primarily on Saturday morning children's programming. Other national accounts captivated by Hummingbird music include Kelloggs, Sunkist, and Goodyear. Hummingbird has also produced gospel albums, and has supplied soundtracks for the CBN cable network.

The growth of Hummingbird Studio parallels the rapid rise of the production company. It started off in 1977 with a "trusty, dusty" ReVox A77, already four-years-old at the time. Although purchased primarily for playback, Farnsworth soon discovered he could make rough demos by pinging tracks back and forth, adding parts until the noise level became objectionable. A second ReVox was added for more flexibility, and soon an embyronic studio formed around a Steinway upright, a Fender Rhodes, four E-V microphones, and a pair of Auratone monitors.

In 1979, the studio moved into high gear with the purchase of an Otari eight-track. Initially, recording and mixing were all done in one room, but soon a primitive (and cramped!) control room was built to provide minimal isolation. With the arrival of a talented and enthusiastic new engineer. Lynn Fuston, the facility quickly became a highly regarded demo studio. By 1981, local clients were accepting demo tracks as masters, a national master (for Putt Putt miniature golf courses) was cut there in 1982, and the fall season promo jingle for Nashville's top-ranked TV station was mastered at Hummingbird in 1983.

The experience gained by Hummingbird staff in this tiny room was instrumental in establishing guidelines for the building of a new 24-track studio. After all, they reasoned, if we can do this well in a cubical closet, then perhaps finicky acoustic design is not all that crucial in making successful commercial music. Or, put succinctly, perhaps State-of-Mind is more important than absolute State-of-the-Art.

Design Criteria for New Studio

The first criterion for the new studio was exceptional comfort. "You have to start with attitude," says Farnsworth. "You want to put people in a place where emotionally they can do their best work. You've *got* to give musicians a space where they can give it their best lick;

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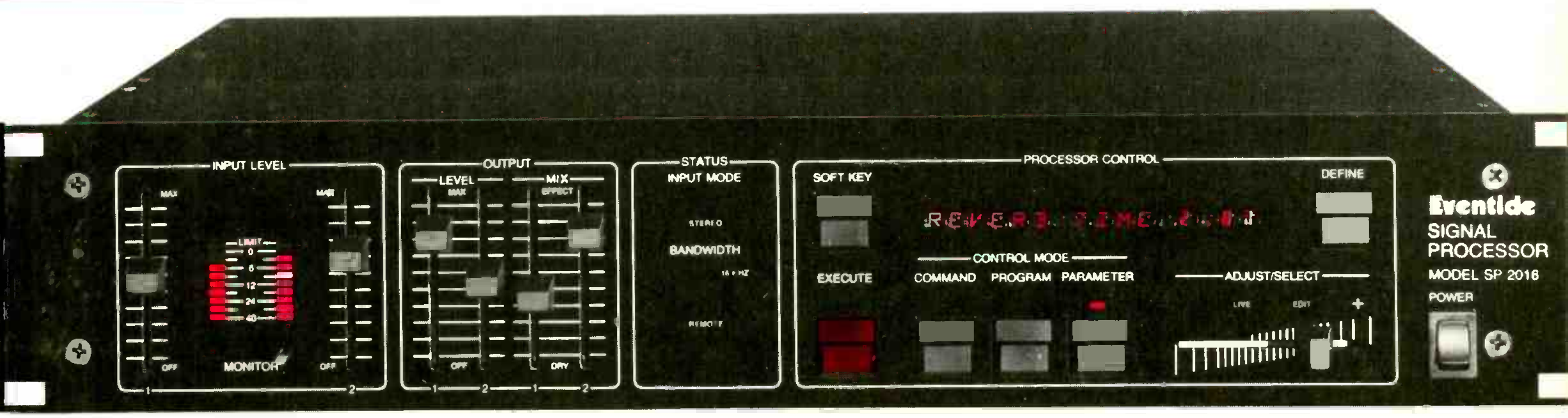

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Communication was also a crucial consideration: Hummingbird producers preferred the immediacy of the small demo studio, and didn't like trying to communicate with singers locked in a booth halfway across a dimly lit, cavernous studio. "It's best to have them right in front of you," Farnsworth insists, "so that if the producer wants to jump up and down with enthusiasm, the singers don't have to *imagine* it over the headphones."

The third determining factor was inspiration. How can you make uplifting music if you are not lifted up? The Hummingbird Studio partners pondered this question, and came up with a literal answer.

A recording studio on the eighth floor in New York is nothing exceptional. In Nashville, it's unheard of! And why should anybody bother, with ground level space relatively cheap?

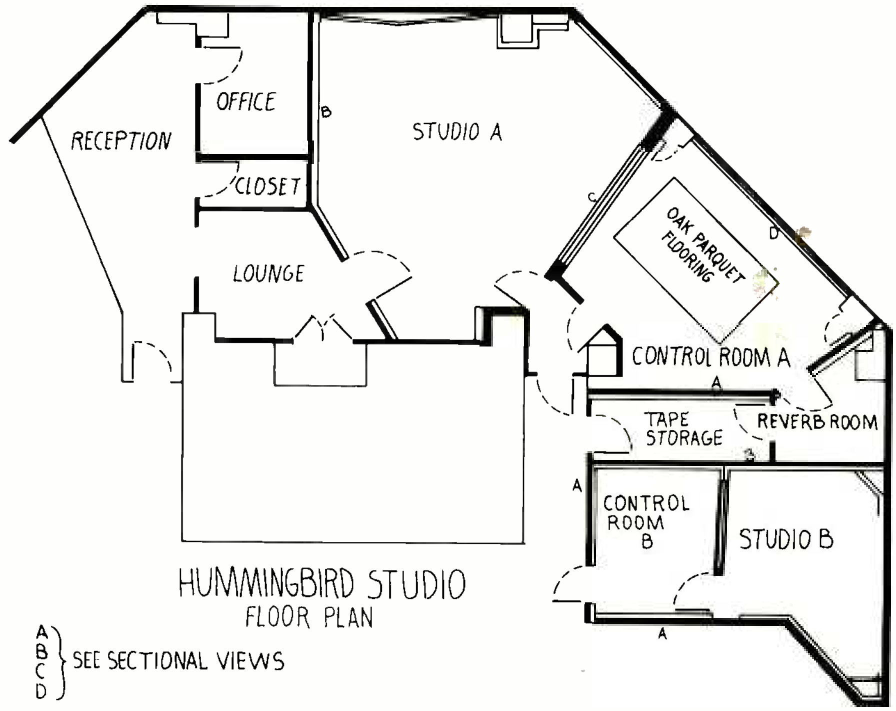
Hummingbird has considered a ground level studio earlier when the company moved its offices out of the United Artists Tower to a smaller building a few blocks down Music Row. But there would be noise and security problems at the new location. So, when they heard that half of the eight floor of the UA Tower would soon be available, Farnsworth, Fuston, and a third partner-to-be, Gary Glover, went up and looked at the view.

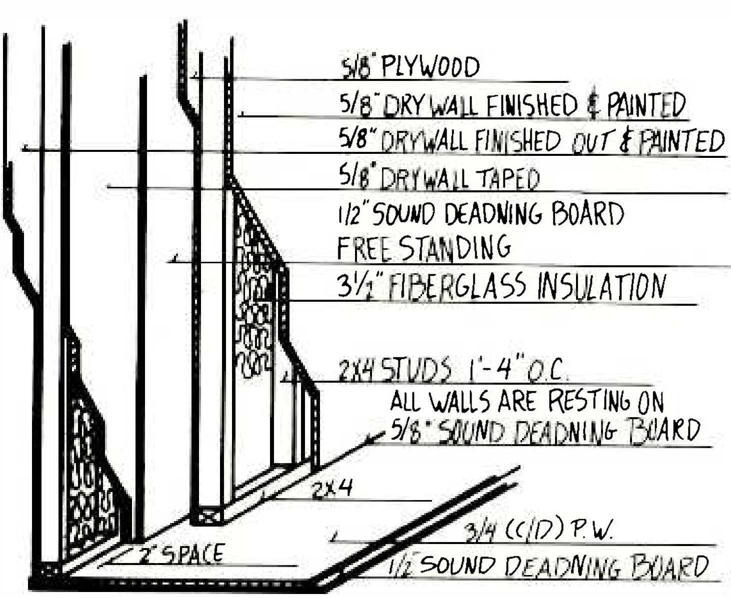
Through two large windows, the view sweeps from south to southwest, from the stately columns of Belmont Mansion to the gothic tower of Scarritt College, across the angular brick and glass ramparts of the new Vanderbilt medical center, then to the classical lines of Vanderbilt's Kirkland Hall spire — all enfolded by the greenery of rolling Tennessee hills.

From the beginning, the studio partners knew there would be problems with the proposed high-rise space. Designed exclusively for offices, the building offered only 11-foot ceilings, with each ceiling slab broken by concrete beams projecting down to 9½feet. Because the building is a hexagon, room shapes would be irregular — for better or worse. And, again for better or worse, half the outside wall area was taken up by windows.

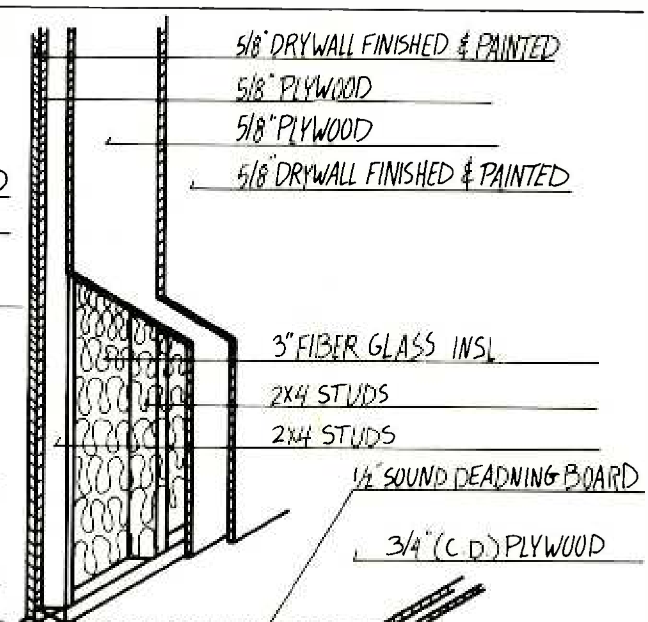
Anticipating problems, chief engineer Lynn Fuston called in acoustic designer Steve Durr as a consultant. Although he quickly realized the building space was, in many ways, "an acoustical nightmare," he decided to go ahead with the job. "When Lynn came to me, I told him there were certain limitations on what I could do because of the building space," says Durr. "But I said I'd do it, knowing it would be a good challenge. I also told him we'd have to work on it over a period of time. We couldn't just throw together some plans, and have it come out perfect. Working with these limitations there are just *too* many variables."

There were also built-in complications. First, the building has no freight elevator, so an outside window would have to be removed to allow deliver of building materials (and the baby grand piano) by crane from the street below. Second, the building's cold water air-conditioning system has individual blower units in the offices. Five units would be located in the space leased by Hummingbird, with two immediately



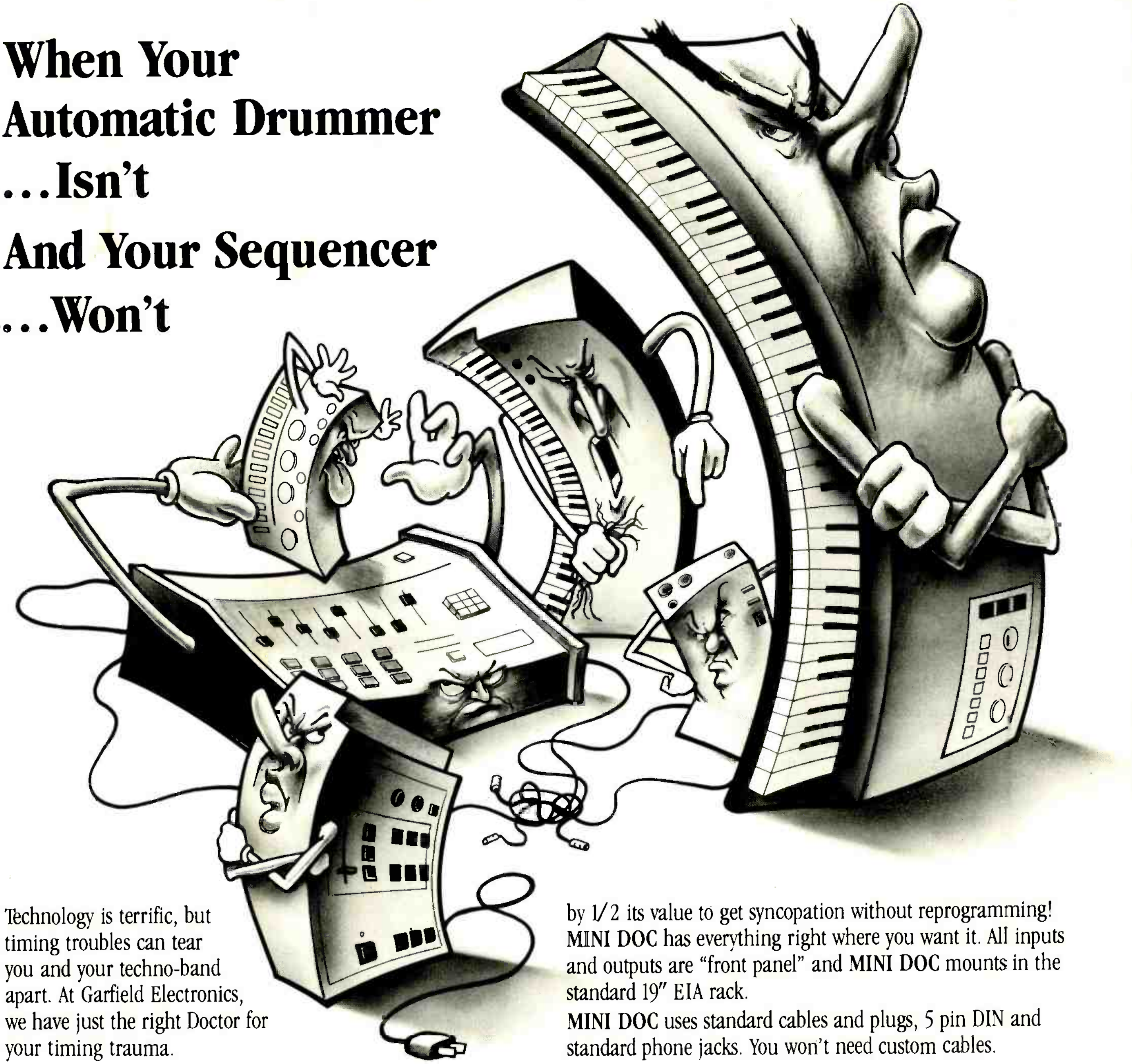


SECTIONAL VIEW A



SECTIONAL VIEW B

When Your Automatic Drummer ...Isn't And Your Sequencer ...Won't



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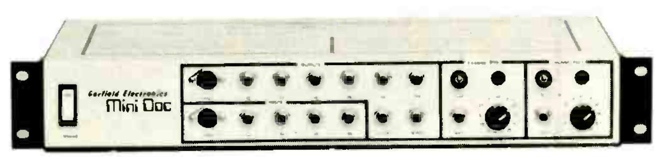

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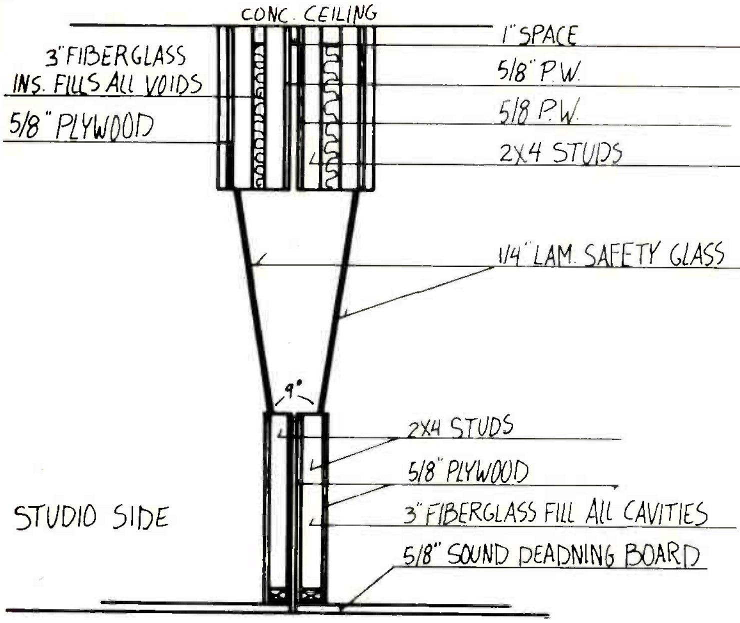
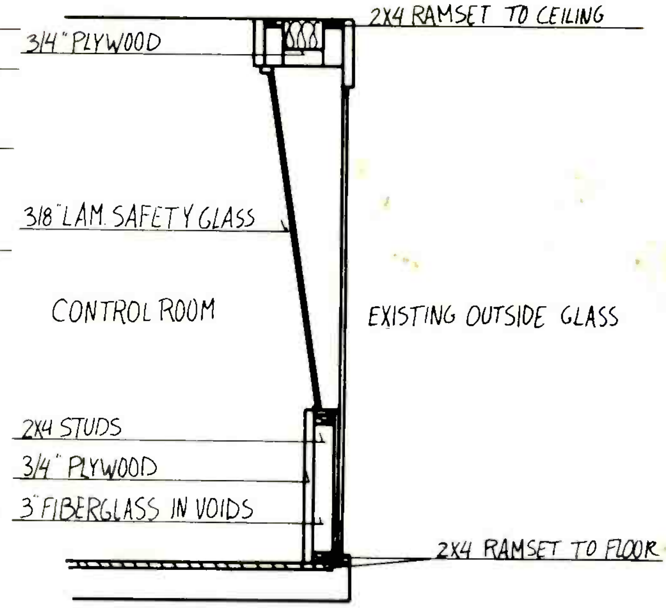


SECTIONAL VIEW C



SECTIONAL VIEW D

adjacent to the studio itself! Third, another recording studio, Suite 900, was located on the next floor up, though thankfully on the other side of the building. And, finally, a different set of fire codes applies when you build anything above the third floor. Hummingbird has a choice: either use metal studs and drywall exclusively (you can imagine how that would sound), or coat every piece of wood thoroughly with fire retardant chemicals. Slop it on, gang!

Undaunted, Fuston and Durr proceeded with their design, working from guidelines Fuston had developed after exhaustive research. Two years before construction got underway, Fuston began making diagrams of all outside studios where Hummingbird producers were cutting masters. Control room diagrams showed shape, console and tape machines location, monitor placement, windows, and seating. Studio diagrams indicated isolation booths, usual location of instruments, and control-room window location.

Later, Fuston circulated a questionnaire among all five staff producers, asking them to list their specific likes and dislikes at the seven studios most often used for by Hummingbird for master sessions. Fuston found the producers wanted a studio with "a homey feel," not a place with "a sterile, antiseptic environment." Comfortable seating with a decent view for clients was important, but not at the expense of the producer and engineer. (Some studios, of course, put a couch between the console and the control room window, thus allowing clients to block the views and, when monitors are mounted low, the high frequencies as well.)

Custom Layout for Jingle Productions

Drawing on these comments and his own personal experience, Fuston designed a studio and control room to satisfy most — if not all — of the producers' demands. The console would be mounted at right angles to the window (a very large one) facing into the studio. A large producer's desk would sit up against the window, and a couch for clients would be behind both work areas, placed so clients could see but still stay out of the way. Tape machines would be off to the side of the console, so engineers would not have to weave through clients to get to them.

All well and good, but how were Fuston and Durr to get this ideal studio into this oddly-shaped building space? Only with great difficulty. "We must have tried 25 or 30 different arrangements," Fuston admits. "And I dare say there has never been a studio or control room shaped like ours!"

"It's pretty much just the way things came out," adds Durr. "First, we had to have a sound lock by the entrance, since we didn't want to open directly into the hall or control room. And we had the air-conditioning unit which put a knot over on the other side. We must have changed the dimensions 25 times. Each time we re-did the plan, I would punch it into my little HP programmable calculator, and it would spit out all the axial modes in the room and tell us if we had a preferred ratio. So we'd try something, run it through, and see if it would work."

In its final design, the bare Studio A (the large room) is very live, with hard wall, a red oak floor, a 10-foot outside window, and a 12-foot control-room window. The ceiling is deadened with #703 compressed fiberglass covered by loosely draped cloth. The room can be selectively deadened by throw rugs, baffles, and hanging panels of fabric covered 703 fiberglass.

Combining the given building space with Fuston's preferred design left little choice for large control-room monitors: there could not be any. The console would have to face the outside window, and the 9½-foot ceiling height at that location left no room for big boxes. Fuston was not concerned. He found most staff producers relied on close monitoring for consistency, and he had used noting else in the old studio — where *everything* was close!

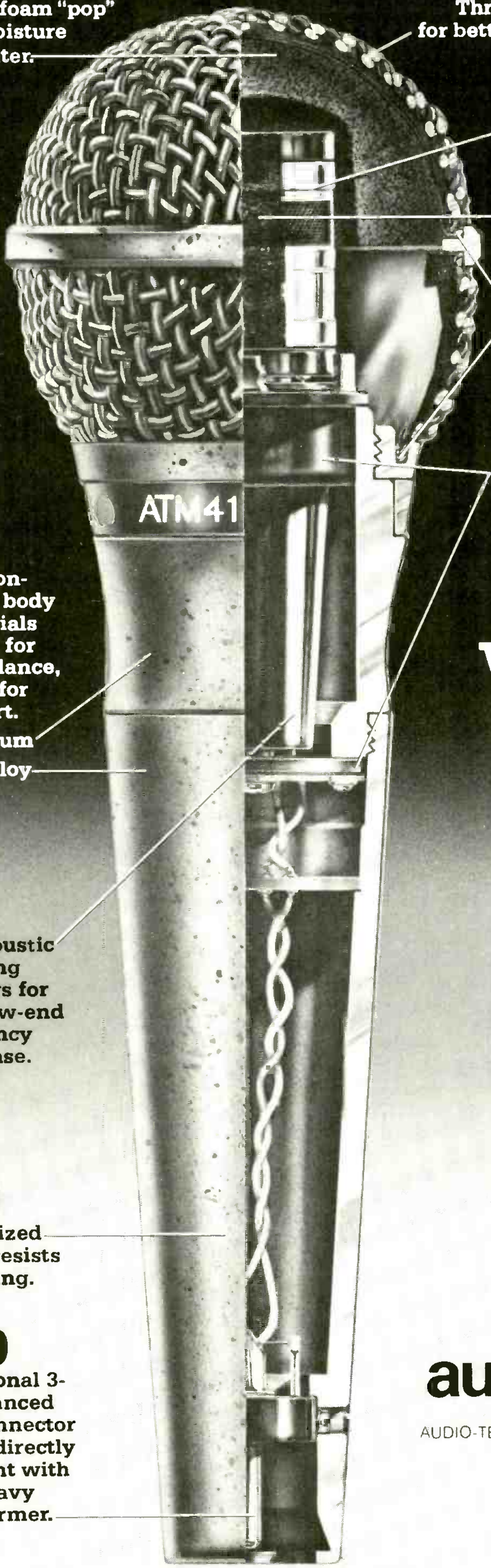
When Fuston feels homesick for the old closet, he can retire to their other studio, which houses the old eight-track equipment. A modified recreation of the old room, it uses a tile floor for liveness and Sonex foam for selective deadening. Two-way video links and patchable

CONTROL-ROOM EQUIPMENT

Studio A:
Trident 28-in/24-out Series 80B console.
Studer A80VU 24-track.
Studer A810 two-track.
Studer B67 two-track.
Yamaha and BGW power amps.
Shure, Neumann, E-V, Sennheiser, AKG, and RCA microphones.
JBL 4311 and Auratone monitor 'speakers.
Outboard: Eventide H949 Harmonizer, two Audio Arts 4200A equalizers, dbx Model 161 compressor, UREI 964 digital metronome, three LP-1 plate reverb systems, live echo chamber (9-storey stairwell).
Instruments: Baldwin baby grand, Oberheim OB-X synthesizer, Yamaha DX7, Fender Rhodes, Krumar Orchestrator, Minimoog, Yamaha Electric Grand, Pearl drums.

Other Studio:
Sound Workshop 12-in/8-out 1280B console.
Otari MX-5050B eight-track with dbx noise reduction.
Studer B67 two-track.
Revox A77 two-track.

Video
Sony V0-2600 ¾-inch U-Matic.
JVC BR7100 VHS VCR.
Sony KV-1922 color monitor.
Ampex AG-440B four-track.



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inputs will allow the smaller adjacent studio to be used as a large isolation booth for sessions in Studio A. A total of 56 inputs are spread through the complex, and all may be patched into the Trident Series 80B console.

Noise Control and Acoustics

Acoustic treatment at Hummingbird revolved around a hornet's nest of noise isolation problems. First, all the outside windows had to be supplemented by a second inner window of ⅜-inch laminated glass. Because of the studio's distance from the street, the only threat of outside noise comes from thunderstorms in the immediate vicinity. Durr shrugs it off: "I learned my trade in a studio where, if it thundered, you quit?"

A knottier problem was presented by the inside air-conditioning units. A heavily insulated closet with a thick hardwood door was built around the one unit in the studio itself, with a maze of fiberglass baffles leading to the unit. All returns had multiple sound isolation baffles built into the ductwork.

Building space stripped prior to studio construction. This view is from the future control-room area through the studio recording area and office space.

The thick concrete ceiling slab and multilayered studio walls eliminated all interference from the studio on the floor above — except for, occassionally, the kick drum, which was coupled directly to the floor. To solve this potential problem, Hummingbird volunteered to build a drum riser for their neighbors.

Control-room acoustics presented a particular challenge to Durr, who is not found of small, close monitoring systems to begin with. "It's much more live than most rooms this size," he says. "I made sure there was enough ambience to allow the sound to carry well beyond the console. That's very important to me, because I try to build rooms so they sound the same, as much as possible, anywhere you are in the room. If you bring the whole rhythm section in for a playback, and it doesn't sound good where they are standing, that can blow the whole energy level right there.

"At the same time, I had to be careful to design the room in a way which — despite all the glass — would still allow you to know how much reverb was on the tape, and still maintain a *true* stereo image."

Fuston admits candidly to some compromises in the monitoring system, but is not greatly concerned: "I'm not going to take somebody into that room and say, as far as response from 20 to 20k, that there's not a flatter room in town, because I know it's not true. But as far as what you hear, it's a balance between the real world, and the sterile studio environment. Yes, it has some quirks, but based on our experience in our old closet studio, we know we can deal with the imperfections. We can adjust to it."

Fortunately, the building space did not force similar stringent limitations on other equipment selections. With a somewhat constrained budget, Fuston decided to go for to quality in his basic, and leave the extras for later. The Trident Series 80B console was selected because "it is the most 'discrete' sounding of the new-chip consoles." Studer A80VU MKII 24-track and A810 two-track machines were chosen for "excel-

PROFILES OF HUMMINGBIRD STUDIO ENGINEER LYNN FUSTON AND DESIGNER STEVEN DURR

Bitten by the recording bug while attending Baylor University in Waco, Texas, **Lynn Fuston** first tried to break into the studio business in nearby Dallas. Unable to get his foot in the door, he made the move to Nashville, where opportunities were more abundant, and where he could get specific training by enrolling in the Music Business Management program at Belmont College. To supplement his classroom education, Fuston sought work in Nashville studios, eventually landing a job as a second engineer at Sound Stage Studios. ("I was hanging around bugging the studio manager every day, day in, day out," he confesses, "so she figured since I was there all the time, she might as well use me!") Later, he picked up more experience working as a second for Travis Turk, a highly regarded Nashville independent engineer at the time.

When Hummingbird expanded from a simple tape-copy operation to an eight-track demo studio in 1979, Fuston's college roommate, the studio's original engineer, felt like he was getting in over his head. After repeatedly seeking advice and assistance from Fuston, the gracious roommate finally stepped aside and gave Fuston the job. By scheduling his Belmont classes during early morning, late evening, and weekend hours, Fuston was able to complete his degree program while working practically full time at Hummingbird.

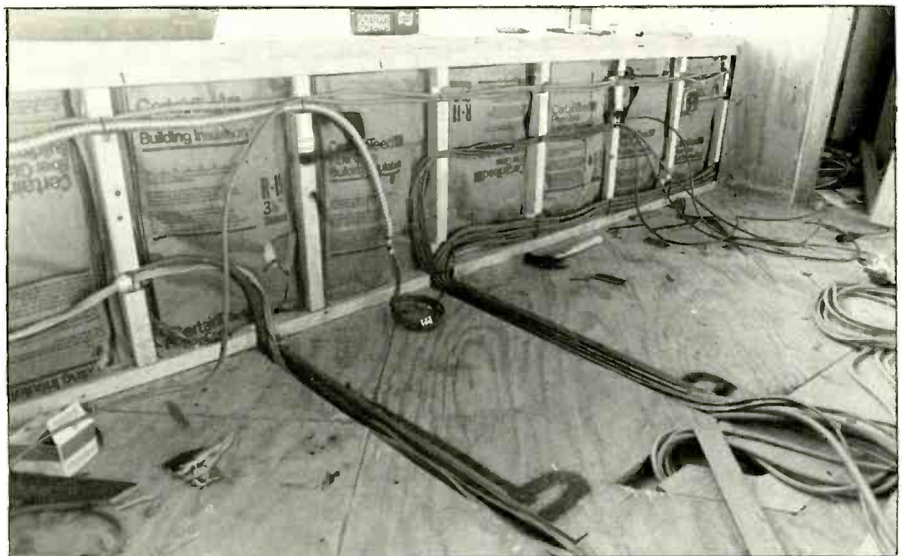
When construction of the new 24-track studio began, Fuston ascended — at the ripe old age of 25 — to the newly created post of vice-president and studio manager, though he continues to engineer many sessions himself.

Acoustic designer **Steven Durr** was thrust into the business back when the Sixties rock explosion first struck his native Baton Rouge, Louisiana. Durr was working next door to the home-town sound contractors that had been hired to supply a PA system for the city's first rock festival. The sound company,, All Technical Industries, asked Durr, one of the few "longhairs" around, to help them deal with these strange "hippie" people from out of town. After the festival, Durr went to work for ATI on a full-time basis, primarily doing commercial sound installations. Later, he turned to PA work for Louisiana bands, and then to engineering in the city's one and only recording studio. He eventually started his own business, but soon found out that Baton Rouge offered "a lot of music, but not much business."

In 1978, he moved to Nashville, establishing Steven Durr and Associates, with his wife, Nancy. For the first few years, the two concentrated on tuning rooms, and their reputation for making bad rooms better, and good rooms great, quickly earned them a steady clientele.

Durr later expanded his business to include monitor-system design and, eventually, complete studio design services. He has designed studios in Pennsylvania, Missouri, and Alabama, as well as in Nashville. He has also done extensive reworking of monitor systems for famed producer Chips Moman's studio as well, as for Ronnie Milsap's Groundstar Laboratory.

Although a graduate of the Syn-Aud-Con seminar series, Durr admits he is not a technically oriented designer. He prefers to rely on his experience, his trained ear, and common sense. His working rule of thumb is, "If it sounds like a loudspeaker, it *isn't* right!" □□□



Cable runs for Hummingbird's Trident Series 80B 28-input/24-group console. The picture window that provides an unobstructed view of the Nashville skyline is seen at top, with console placement area designated to the lower right.

lent overall performance, and reliability." Outboard effects are limited, but are available on rental at short notice; an accompanying sidebar lists the facility's main equipment.

For most occasions, reverberation is supplied by three plate units designed by Gene Lawson, and distributed by Blevins Audio. But for special applications, Hummingbird has possibly one of the most mind-boggling echo chambers ever heard on tape: a nine-storey concrete stairwell. Loudspeaker and microphone lines are always at ready must inside the doorway, so the chamber can be "activated" — preferably after regular office hours — at a moment's notice.

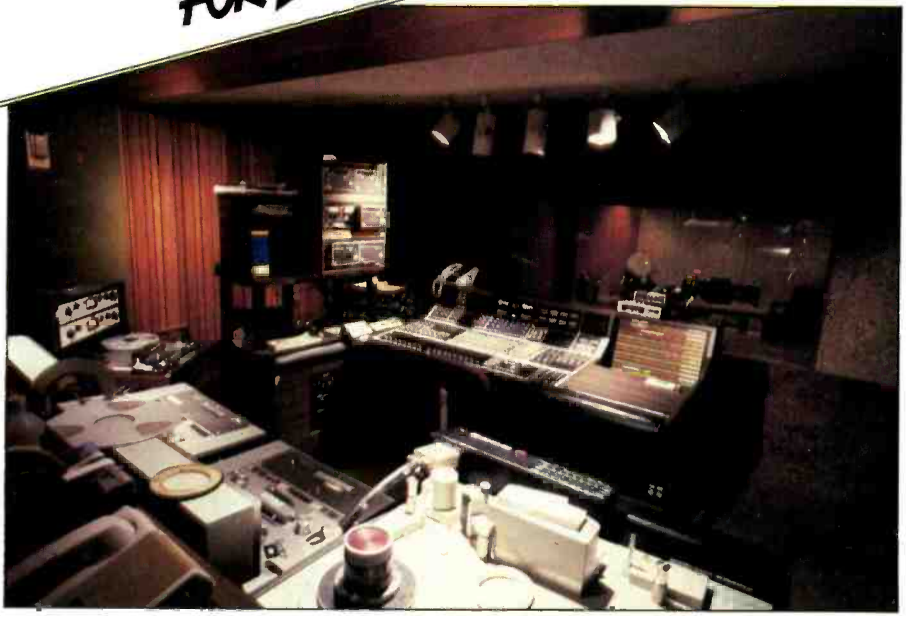
"The sound is just *awesome*," says Fuston, "but you have to be careful because sometimes the security guards will come through. You might not notice it right off if the reverb is down in the mix, except at the very end when you'll hear coughing, and the clunk of a door closing. Once we were testing the stairwell by hitting a tom-tom a couple times. There was a security guard down in the lobby, and he ran up eight flights of stairs. He thought a gun had gone off!"

Client Reaction

Although the new Hummingbird Studio had been open for only two weeks at the time of this writing, initial client response had been very positive, the owners say. The first sessions evoked praise for both the sound and the smooth operation of the facility. Nevertheless, given the novel location, you can't help but wonder about the distraction factor.

Case in point, from the first week of operation: Session concluded, final mix played back. Engineer and producer, both pleased, look up at the client (hailing from the flatlands of Texas), who is grinning ear to ear. "What do you think?" the producer asks. "It's great!" the smiling client responds. "It's really an incredible view!"

Deciding to have a studio "lifted up on high" simply for the sake of a view may seem risky, even foolhardy. But the folks at Hummingbird, perhaps sustained by their common Christian commitment, never gave the risks a second thought. After all, when it comes to music (commercial or otherwise), a little inspiration can go a long way. ■■



Photography by Kathy Cotter

MULTITRACK BROADCAST PRODUCTION AT ABC/WATERMARK

by Adrian Zarin

"Multi-track production is the *only* way to do a lot of the shows we produce," reflects Peter Skye, chief engineer for ABC/Watermark, the facility that creates such popular weekly radio programs as *American Top 40* with Casey Kasem, *American Country Countdown*, *Soundtrack of the Sixties*, and *Soap Talk*. It also produces several multipart specials, including the 26-hour *Ringo's Yellow Submarine* with Ringo Starr, not to mention *Spotlight Specials* on artists such as Linda Ronstadt, Air Supply, and Rick Springfield. Previous productons have included the science-fiction drama serial *Alien Worlds*, and the 13-hour *Elvis Presley Story*. In virtually all of Watermark's productions, Skye asserts, multitrack recording techniques and equipment play an indispensable role.

"For a show like *Soundtrack of the Sixties*," he offers, "you might want to play a series of hits a certain artist had, or to play just a section of each record. But rather than just assembling a montage [by] stringing them all together, you might want to have an announcer give the title and year for each song, and then drop in the musical pieces. Along with that, you could have a bed of background music, say the instrumental passages from the artist's biggest song. This would be dropped out whenever another piece of music came in, but would always be under the announcer. At the same time, you might even want to have sound effects. For an artist who is associated with the great outdoors, like John Denver for example, you might have crickets chirping in the background.

"You're cueing up five to 10 different elements there. It would be *impossible* to do the show with a stack of two-tracks. There would be too many buttons to be hit at just the right time. So you're pretty much locked into multitrack production."

Specialist Production Requirements

The ABC/Watermark facility includes four production control rooms, two of which have voice/music recording studios attached and are equipped with 16-track tape machines. While the equipment in all the rooms has been modified to meet Watermark's specific requirements for multitrack radio production, the API console in Studio A has undergone particularly extensive modification. "It's a combination of the API 1604 and API 2488," Skye explains. "We customized it from scratch. I went to the factory on Long Island where they were building it, and we did the modifications while the console was being built, which was close to 10 years ago." Among the custom features are the board's input modules: each input has 15 possible destinations (plus a solo feed) that perform a variety of functions.

"On the input channels, I needed a monaural cueing function with the fader fully down," Skye continues. "You don't need a volume control, but you need to be able to hear the record as you're back-cueing it [to locate the precise point where the music starts]. The essential thing is to get the record cued up as fast as possible.

"In addition to this, I needed the monaural cueing function to be switch selectable, so that we could shut off a channel during a multitrack mix without the sound coming out of the cue monitor."

Each input module is also equipped with a stereo audition cue send. As Skye explains, "A stereo audition switch was added to the stereo foldback feed on each input, so the engineer could rehearse the next 'break' while a record was playing. While you've got one record going to tape, you're rehearsing the next record —playing the intro and dropping in sound effects, the announcer or whatever — in order to get the timing pefect. Naturally, you don't want to record that sound; you just want it to come through the monitors, which is why we need the audition feed." Full metering facilities enable an engineer to simultaneously monitor Program and Audition levels, as well as limiter send/receive, etc.

Skye also had 48-volt phantom power added to each input module, and which can be switched on and off as needed. "You need the power for the condenser mikes," he explains. "But some dynamic microphones have a leakage problem when 48V is in use. The switch eliminates the noise. Some consoles today have this feature built in; 10 years ago they didn't."

A further modification was to permanently wire UREI 1178 stereo limiters into each of the console's submasters. According to Skye, "They function as submaster line amplifiers, and have been calibrated to each console's gain structure. We then remove the output level knobs, and lock the controls."

Valley People Gain Brain compressor-limiters and Kepex noise gates have also been built into the board, and are accessible via the patchbay. The built-in patchbay is rather extensive, and contains a jack at every function point, including Fader Out.

"I want to be able to solve session problems *fast* by locating the trouble and patching around it," explains Skye. "Also every VU meter is patchable, which increases the board's flexibility and speeds up meter calibration."

As for the console's output structure, the engineer continues, "there are simultaneous stereo and monaural outputs [monitored via three VU meters]. The output bus drives a monitor oscilloscope located between the two monitor speakers; it basically lets you see what the program is going to look like. Also, you can spot phasing problems, channel imbalance . . . things like that."

Further modifications include a

PRODUCTION

Heightened consumer awareness, the success of music video, and the resurgence of live programming challenge broadcasters to achieve new standards of audio performance. The quality of new source material, such as the Compact Disk, places exceptional demands on the audio chain. Production consoles must meet these demands as well as provide systems unique to the broadcast environment.

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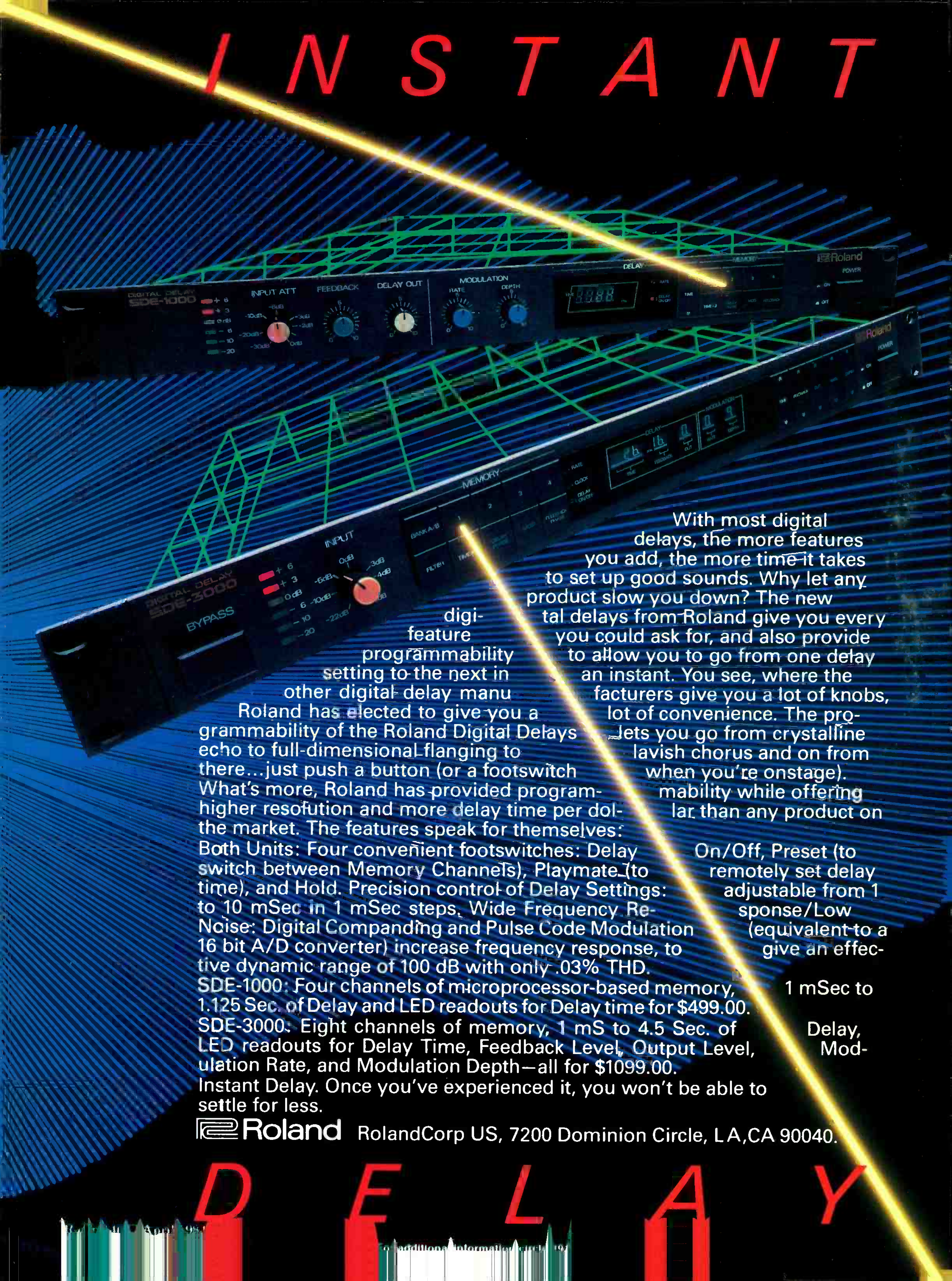
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console-mounted remote panel for controlling stop/start and switching functions on the studio's turntables, reel-to-reel and cartridge machines, Dolby noise reduction units, and clocks. "Also, there is a custom-built hybrid telephone hookup which enables an announcer or artist in another city to literally phone in his part, or a producer to oversee a session from out of town."

Other consoles around the facility include Quad Eight and McCurdy models, as well as a new 40/48 Harrison MR2.

The many sound sources that go into the making of ABC/Watermark's radio programs are stored on a variety of media. Apart from a Dolby-equipped Ampex MM-1200 with interchangeable 8/16-track headblocks, the studio has two four-track, half-inch machines (an MCI JH-110, Scully 280, and Studer B-67) and 16 two-track, ¼-inch decks (an assortment of MCI, Studer and Scully transports). Why so many tape machines? "So we can source all the materials that are being picked up for production," Skye responds. "For example, if we're using just brief sections from a number of songs, we'll do a 15 IPS dub from the record, edit the dub, and then use that for production. We might splice each selection into a loop, and just put a leader on it. We might have three or four machines loaded with music drops and beds, and another one with sound effects.

"Now, we *could* handle all of these with just one two-track going to a multitrack, but it's a lot faster and easier just to load up several tape machines than it is to be constantly swapping reels on one machine. It costs around $10,000 to buy a tape machine, but in six months' time it's going to save you at *least* that much in studio time, man hours, etc."

Along with the tape machines, source material for dubbing to multitrack is also available on records and cart machines — two staples of radio production. The studio uses Technics SP-15 turntables with pre-amps by Shure, Audio-Metrics and Bill Isenberg, Technics EPA-100 and EPA-A501M tonearms, and Stanton cartridges with conical stylii. "Elliptical stylii

— Chief Engineer Peter Skye —

cut up the records too much when you back-cue," Skye explains. KLH TNE7000A Transient Noise Eliminators and DNF1201A Dynamic Noise Filters also are used as needed to clean up the sound quality of disks used in shows.

Rounding out its complement of control room sound sources, Watermark has 24 ITC Series 99 cart machines (in which the facility use Aristocarts), along with nine Technics and four Sony cassette decks.

Control Room Design and Layout

With so much equipment on hand, a lot of thought naturally went into control room design and layout, as Skye explains: "The most unique thing that needed to be designed into our control rooms was space for the *large* number of two-track tape machines we use. We did drawings of the equipment racks to make sure everything would fit. The turntables are always located to the left of the console, rather than on the right, since the tonearms are then closer to the front and therefore easier to cue. The turntables are mounted in their own cabinets. In our main studios, the cabinets are on casters so they can be rolled around. The bases are properly isolated from the cabinets to minimize mechanical transmission noise. The cabinets themselves were internally reinforced, because they tended to resonate at higher monitor levels.

"All of the cart machines are mounted above the turntables with proper mechanical isolation so that motor noise isn't picked up by the turntable cartridges. Isolation materials we use include heavy wood panels, sand bags in each compartment of the cabinets, industrial shock mounting, and machine rubber."

Acoustic design features in the control room include a compression ceiling. It slopes forward from the control room glass and is lowest at the mixing position, which serves to direct the sound coming from the monitor system. "It's like a giant megaphone," says Skye. "There is more sound rpessure level at the mixing position. It does affect your frequency response a bit, but it makes the sound louder, so it's a trade off."

Monitoring in Watermark's two large multitrack control rooms is provided by UREI Time Align loudspeaker systems, which have been removed from their original cabinets and custom mounted into soffits in the control-room walls. There is also a generous complement of JBL 4310, 4311, and 4315 monitors scattered throughout the facility. All monitors are powered by Crown amplification, and room EQ is provided by UREI third-octave room equalizers.

Studio Acoustics

While a great deal of Watermark programming originates in the control room, the focal point of many of its productions is a "live" radio announcer/personality, such as Casey Kasem, Bob Kingsley, or Gary Owens. Production on these shows is handled in the facility's two large studios — Studios A and B — each of which features its own voice booth/music room.

"The rooms are designed to be large enough for in-session production meetings between the talent, producers, and writers," Skye notes. "There will often be six people in these meetings."

Room acoustics in Watermark's studios were designed by acoustic consultant George Augspurger to handle music, as well as voice sessions. Sound traps are located at either side of the room. No two walls are parallel, and acoustic surfacing materials include fiberglass panels

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Studios D (left) and C feature identical McCurdy production consoles equipped with 24 stereo input channels. Available tape machines include MCI JH-110 two- and four-tracks, Scully 280 and SP14 two-tracks, and ITC Series 99 cartridge transports. Turntables are Russco and direct-drive Technics.

tages — not the least of which is that while studios in most cities dispatch tapes to out-of-town clients by taking them down to the Post Office, Video Post and Omega personnel can literally walk over to the Federal Express terminal, and watch their cargo get loaded on an airplane. It also means ready access to clientele within the wide region served by Southwest Airlines and Muse Air, both of whom land at Love Field.

Dallas Sound Lab is part of the Dallas Communications Complex — a.k.a. The Studios at Las Colinas — located in suburban Irving, the first stage of which opened last summer. Besides an audio studio, the complex, which resembles an ultra-modern college campus, includes Dallas Post-Production Center, Victor Duncan equipment sales and rentals, and three complete soundstages that are owned by the complex itself and leased to production companies. There is also in excess of 300,000 square feet of office and warehouse space, which eventually will be occupied by editing labs, production companies, casting and advertising agencies, carpentry shops, law firms, and all of the other businesses essential to film and video production. The plans for the complex are grandiose; the existing buildings represent only one-tenth of the proposed construction of the site.

Main control room at Omega Audio, designed for 24/46-track recording, video sweetening, and mixdown. Equipment includes an AMEK 2500 console, and Otari MTR-90 24- and MTR-10 two-track machines.

The New Hollywood?

According to some, Las Colinas was originally designed to lure film producers away from Hollywood. Says John Marshall, manager of Dallas Sound Lab, "This is a big location-work area, so if we offer soundstages and posting facilities, maybe they'll do it all," especially if such facilities are made available at attractive prices. There are even stories of the complex' promoters holding huge "Texas-style" barbecues in Tinseltown to help spread the word.

But today there is general agreement that this strategy is going to have to be modified. "I believe it's valid to say that if you're shooting in Texas, if you want to do some indoor stuff while you're here, we've got the soundstages," Christensen says. "But that's very different from somebody literally picking up from LA and coming all the way here to do their interior shots and then hanging around to do their posting. Why should a producer or director stay here for six months, when he can stay in Hollywood and drive home every night? These guys aren't interested in saving money; they're interested in saving their mental and physical health."

"It's hard to convince LA people to stay here to do scoring and posting," Marshall admits. "They'll always want to go back to LA or New York for scoring — they're used to working with certain composers and arrangers, and they won't fly them here."

Interior View of Omega Audio's 24/46-track mobile facility, utilized for in-field production and multitrack recording with video.

Building Regional Business

Nevertheless, there seems to be plenty of specialist audio work to go around — but it is coming from the region surrounding Texas, stretching as far as Nevada and Mississippi, and going down into Mexico. And the sessions are varied: TV and radio commercials, industrial and educational videos, custom records, political campaigns, religious presentations, music videos, and network, independent, and cable television programs.

Facilities in Dallas are learning new ways to attract and keep business. At Love Field, clients are presented with an

Omega Audio's studio recording area provides variable acoustics and moveable screens to create individual recording environments during audio sweetening session.

COST-EFFECTIVE AUDIO-FOR-VIDEO AT WPCB-TV

The two units are similar in features, and work very well together. Both consoles are bussed together through the eight sub-masters for dual tracking. The M-50's frame holds 12 input modules, joined in groups of four for removal, thereby providing a total of 36 available inputs.

The M-50 has several features that would be useful if they were available on the M-16. For example, balanced outputs are provided on the M-50, switchable between a +4 and a +8 dBv, depending upon your system requirements. As it turned out, the M-16's main output was run through a balancing amplifier to make the system compatible with the equipment downstream.

The M-50 also has cut or mute switches on the cue sends that are particularly handy when pre-setting cue or foldback levels to musicians in the studio. Both consoles have switchable pre- and post-fader echo and cue sends.

Actual construction of the control room was done in two stages, so that regular productions could continue during the upgrade. The front half of the room, which was finished first, contains the main monitor cluster that supports both audio and video monitors. It has always seemed to me that audio-for-video installations are usually lacking in either good audio monitoring, or sufficient video monitors to view all the necessary picture sources during a production or editing session.

JBL 4430 monitors were installed because of their phase accuracy and flat frequency response, and are driven by a Crown PS-400 amplifier. Directly above the amplifier — which is located within the monitor cluster enclosure — is an LED meter that measures its output power; it is most useful for assuring that equal power levels are being sent to both speakers as we mix in mono or stereo.

Located between the two JBL monitors are a pair of color video monitors displaying Preview and Program video outputs. The Preview monitor can be switched to provide a view into the studio through any of the station's four video cameras, as well as looping off the director's preview video bus from the switcher so that we can see how a shot or effect is being set up. Also located in the cluster are a pair of Auratones, a Crown D-75 to drive them, and a TFT meter that monitors the actual aural output from the TV transmitter. With the help of Jack Girty, a very creative carpenter in the area, we assembled the design of the monitor cluster enclosure, finishing the exterior surfaces with gray Vermont barn board. Almost all of the other carpentry work was done in-house.

Three independent lighting circuits were utilized: one for track lighting above the console; another for peripheral indirect lighting; and a third for a fluorescent

View from mixing position: TEAC/Tascam M-16 24-input/eight-group console, and JBL 4430 monitors. Top picture is the rest of the control-room hardware, including a Tascam M-50 12-input auxiliary mixer, Ampex ATR-700 stereo transport, dbx 163 limiters, UREI LN 1176 limiter, Bimap reverb and Ampro-Scully cartridge decks.

fixture only used for maintenence. In addition, trim lighting was used around the base of the console, as well as the front of the audio racks. Not only did the lighting make the installation look more "expensive," it also prevented glare from the faces of the TV monitors. All incandescent circuits are dimmable.

To keep all wiring concealed as much as it is practical, a special cabinet was designed to the left of the console. Also inside the cabinet are to be found electrical outlets for test equipment, and two small light fixtures for trim lighting around the base of the console.

The main rack located behind the console holds all patching, switching, cart storage, and support gear. In order to reduce background and unwanted ambient noise produced in the TV studio itself, we are currently using two Symetrix SG200 dual noise gates, which interface well with the M-16. The degree of gating can be set on the SG200s so that they will let a little sound leak through when they are "closed," so that an undesirable "vacuum" effect doesn't occur. The noise gates are genuinely helpful on talk shows using multiple mikes when guests will talk out of order unexpectedly. Also in this rack are two dbx Model 163 limiter/compressors, a UREI LN1176 limiter, a Biamp reverberation unit, and an

. . . continued overleaf —

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Ampex ATR-700 two-track.

A third rack, to the right of the console, holds the Tascam M-16 power supply, an RTS Systems intercom, two Ampro-Scully NAB cart machines, and a TEAC cassette recorder.

Out in the studio audio and video recording area, there needed to be microphone terminations in several places so that sets positioned anywhere could be serviced. Since nobody likes to see hundreds of patch cords hanging out of the racks, and in order to utilize our three proposed 24-input mike panels, this writer designed a 72-by-24 microphone switching matrix to feed the M-16's inputs. (Inputs to the M-50 would be patchable, but that did not create a problem.) The switching matrix was designed around a set of 24 custom assembled, three-position Centralab switches with silver contacts. Each of the mike panels in the studio terminate in a high density Winchester connector with gold contacts. Two 24-input snakes were designed to interface with these connectors, so that the snake boxes could be brought up close to the sets for neat mike cable runs.

New microphones also were purchased. On the talk set, the facility is now using AKG C451 condensers with tubes and swivels. Other mikes selected include Shure SM-57s, SM-77s, a Sennheiser MKH-416 for location work, some Crown PZMs, and an HME wireless System 85.

The foldback system provided in the studio consists of two JBL 4602 floor slopes for the musicians, and two other systems to handle PA for the audience and talk set areas. Levels are monitored by audio floor personel during a shoot to prevent excessive studio levels that could disturb the quality of an on-air mix.

The author's "Attic" personal-use studio, where the majority of WPCB's original music, including themes, backgrounds, promo spots and station IDs are produced. Equipment includes a Tascam M30 mixer and Model 38 eight-track on half-inch machine. Top is station's original production studio, based on Tascam Model 5 and 5X consoles.

Towards the Future

Future plans include the purchase of a Tascam 85-16 one-inch 16-track, and a BTX or Audio Kinetics SMPTE synchronizer to interface audio transports with the videotape equipment. Simple multitrack recording is now accomplished on a Tascam A3440 four-track on quarter-inch. The area of audio sweetening for broadcast is wide open. Usually this type of work is done only at major production houses, but it is now very feasible for much of this work to be done at the local facility. Since WPCB-TV produces its own programming, the creative potential of this type of system to us is enormous.

At the present time, all of the station's music is produced at my personal-use studio, RS-2. I use an eight-track system centered around a Tascam Model M30 mixer and Tascam Model 38 half-inch multitrack, linked to multiple keyboards, two videocassette recorders for viewing material, some effects, and an MXR Digital Drum Computer. Between Tom Green, a writer/musician at TV 40, and myself, we are a two-man band. Tom plays all guitar parts —electic and acoustic — as well as trumpet and flugalhorn. I handle all keyboards, drum programming and engineering. All of WPCB's theme music, backgrounds, spots and station IDs are produced in this fashion. It's been great not having to rely on "needle drop" or library music.

Currently, WPCB is producing several new shows. One in particular, produced by Tom Green, and called "Lightmusic," comprises a music show based on a Fifties-style recording studio set that focuses on musicans and musical topics.

As a result of all the new programming, we have been forced to re-evaluate the area of downstream audio, or what's in-between the control room and the transmitter. We are working, as funds permit, to improve the quality in this area as well. For example, the station is now in the process of installing a new Bosch audio/-video routing switcher that will help eliminate some audio noise and crosstalk problems we have been experiencing. Also, an Audio + Design/Calrec Transdynamic Tri-Band Processor was recently installed in our main program line to control the level of our broadcast signal. (We were experiencing a loss of fidelity and poor dynamic control with our old system.) A Transdynamic system was ordered with stereo limiters so that when a format for Stereo TV is decided upon, we will be ready. ■■■

CUTTING THE CABLE

tion. Each installation will have its own special problems and its corresponding solutions.

Taking a show on the road and using wireless microphones is a job that would frighten most engineers. However, the system chosen by the road company of *Dream Girls* is an example of a highly portable, yet completely effective equipment package. The production carries five Sony transmitters and receivers, and only one antenna with a splitter and booster. In this case, reliable operation is achieved without diversity reception.

The choice of configuration for a wireless microphone installation is a blend of science and art. Experimentation, improvisation and an understanding of the limitations are the keys to a wireless success.

The Wireless Future

To simplify wireless operation, manufacturers are currently moving into a new area of development: frequency synthesis. Until now, sound companies have had to carry huge stocks of alternate crystal-locked transmitter/receiver combinations in order to guarantee top sevice. With a frequency agile system, however, it is a simple matter to tune in any desired operating frequency.

If we consider diversity reception the second stage of wireless evolution, then frequency synthesis is the third; Sony will introduce the first complete frequency synthesizing system this year. If you've been skeptical about using wireless microphones in the past, now might be the time to take another look at the wide range of possibilities.

Complete frequency synthesis means flexibility at both transmitter and receiver. The stability of crystal control is retained in order to generate a steady signal. A master oscillator is controlled by the crystal, and the generated frequency is processed, and then multiplied to produce the desired transmission and reception frequency. The result is the stability derived from crystal control, together with the flexibility of a tuneable system.

Frequency synthesis is neither magical, nor new to the communications industry. The problem was designing a system small enough to fit into a hand-held microphone or body-pack transmitter.

The impact of Sony's VHF system should be felt immediately by rental companies specializing in wireless systems. Requests for numberous wireless microphones in simultaneous operation are common. Until now, however, rental companies have had to keep a large supply of wireless systems in stock to guarantee reliable installations. The present VHF system was designed to end these headaches. To illustrate the advantage of frequency synthesis, let's consider a typical performance situation. The producer tells his audio engineer he wants 10 wireless systems. The engineer either rents or buys 10 wireless microphones and 10 receivers, with each transmitter/receiver combination operating on a separate frequency.

During the sound check it is discovered that three of the frequencies are picking up interference from other RF sources — perhaps from a theater nearby, or maybe from electric motors operating in the vicinity. Beyond that, there may be conflicts among the 10 frequencies selected.

The engineer is beginning to sweat because the curtain is going up in two hours. An urgent phone call to the rental house produces three new systems. Their operating frequencies are clear but, as a result of the substitution, new conflicts may have been introduced into the system as a whole. What works on the drawing board or in the rental shop may not work in the theater or concert hall.

Conventional wireless systems also create special problems for touring productions. As we've seen, professional VHF wireless microphones operate in the 174 to 216 MHz range, the same transmission band as televison stations 7 through 13. Since television channel assignments differ by market, and the FCC permits wireless operation only in *unused* portions of the band, no single VHF system is appropriate for all areas of the country. Once again, frequency synthesis is the obvious solution to the problem of having alternate systems on hand for the various cities visited.

The major components of he new system are the WRT-210 microphone transmitter, the WRT-220 pack transmitter (with supplied lavalier microphone), and the WRR-210 and WRR-220 receivers. The frequency range of each transmitter covers two adjacent television channels, or 48 individual transmitting frequencies. Thus only four transmitters are needed to cover the entire range between 174 and 216 MHz. The transmitters operate with the following channel assignments: 1. channels 7 and 8; 2. channels 9 and 10; 3. channels 11 and 12; and 4. channels 12 and 13. The assignment scheme ensures availability of operating frequencies in the various television markets.

The WRR-210 and WRR-220 frequency synthesizing receivers complete the basic system; each covers the entire range of frequencies between 174 and 216 MHz, a total of 168 available channels. A simple pushbutton system with LCD readout allows easy tuning to

WIRELESS MICROPHONE CASE STUDY:
RADIO CITY MUSIC HALL, NEW YORK CITY

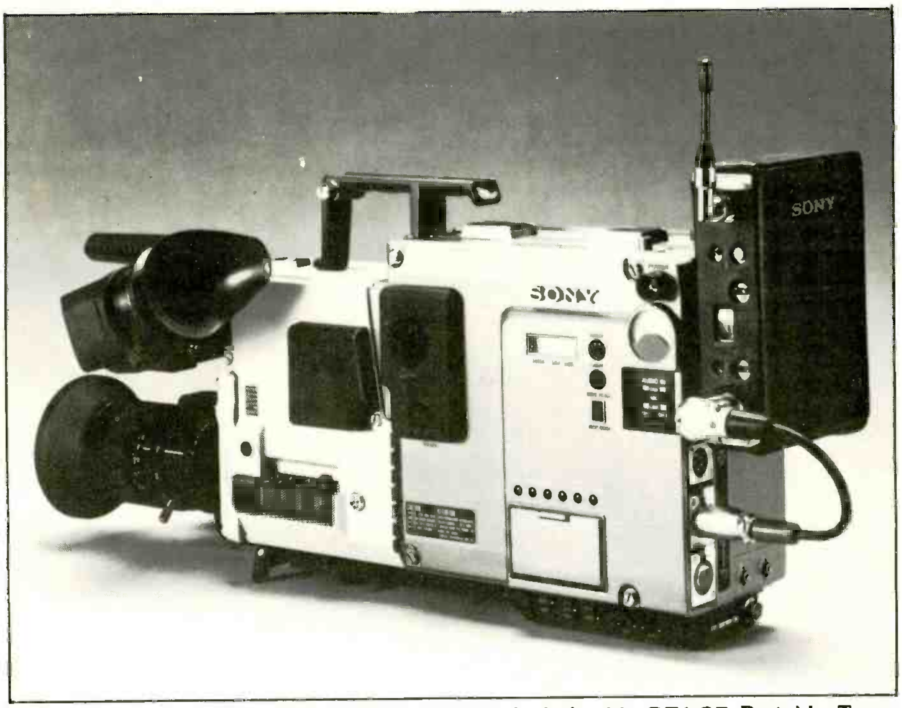
Since its doors opened in 1932 in Manhattan, Radio City Music Hall has been not only an important entertainment venue, but also a showplace of technology. The famous theatre has had a long tradition of innovation in electrical and mechanical design. Today, Radio City Music Hall features a complete UHF wireless microphone system from Sony that offers both "invisible" lavalier microphones, and hand-held microphones with built-in transmitters.

The wireless-mike system was used extensively during the Music Hall's production last year of *Five, Six, Seven, Eight, Dance*, starring Sandy Duncan.

"For the Radio City Music Hall, the age of the wireless microphone has definitely arrived," comments Eddie Santini, chief audio engineer, and designer of sound reinforcement for all productions. "Our productions have a lot of singers and dancers, and they really can't use microphones with cables because they might trip."

According to Santini, the company is using more wireless microphones than ever before. To meet the demand, it recently expanded its equipment complement, adding six Sony WRT-57 hand-held wireless microphones, six WRR-57 UHF tuners, and WRT-2AA body pack transmitters, supplied with ECM-50 lavalier mikes.

The advantages of working without the restrictions of microphone cables are obvious, but until recently wireless systems lacked wide popularity. According to Santini, the Sony system is the first wireless microphone to eliminate dropouts in reception, and extraneous noise interference. "Using only one antenna offstage," he says, "the system delivers perfect reception all the time, despite the cavernous size of the stage. No matter where we go on stage or in the house there's not one dropout." □□□



Typical applications for UHF wireless systems include this BTA-27 Portable Tuner Attachment Kit for Betacam combined half-inch VCR and camera units, and designed to provide cable-free audio links during ENG/EFP video shoots.

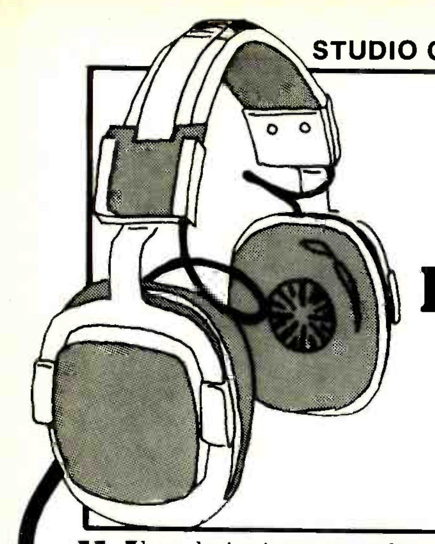
desired frequencies, and retuning if interference is encountered.

The use of multiple wireless systems in simultaneous operation will always be a potential hazard. The new system addresses the problem of frequency planning with a built-in scheme for channel selection. Computer analysis has been utilized to select the most compatible groups of transmission frequencies. Each receiver has been programmed with this information, and will recall via computer memory the compatible frequencies once a group has been selected.

The new VHF system also offers the choice of diversity reception. By utilizing two tuning sections on the same frequency, and a "smart" switcher that samples both inputs. The WRR-220 receiver is able to assure uninterrupted transmission.

Reliability, flexibility, and compact size were sought after by the engineers involved with designing the new system. However, one crucial element must be mentioned — audio quality. Without smooth frequency response and ample dynamic range, *no* wireless system can claim professional lineage. In the past, many users have praised Sony wireless systems for their audio performance; it is expected that the new VHF microphones maintain this tradition.

The introduction of the VHF system from Sony is considered by the company to represent an important landmark in wireless microphone technology. Large facilities employing multichannel systems will enjoy a new degree of freedom and flexibility. Musicians and performers on tour will no longer have to transport alternate systems for the various cities they visit. And rental companies seeking to offer numerous systems will be able to do so with a minimum of inventory while providing off-the-shelf reliability and clean frequency availability. ■■■



DESIGNING HEADPHONE DISTRIBUTION SYSTEMS

by Rick Simon
Simon Systems

When designing a recording studio or related audio system, a lot of obvious questions arise: Which tape deck? Which console? Which monitors? But what about the headphone distribution system? While headphones may not be the most colorful item in the studio, they have a very important purpose. And if you want to put your studio together correctly (don't we all), paying a little more attention to proper headphone distribution at the design stage can really avoid problems down the road.

There seems to be some confusion on the proper methods for driving headphones, which brings up questions like: "Why can't I get enough volume out of my 'phones? How many 'phones can I hook up to my amp? How much power do I need? Should I use active or passive distribution boxes?" It is my intention to provide some insight that hopefully will answer these and other questions, and then suggest some easy "Rules of Thumb" that can be used as a reference in designing a proper headphone distribution system.

Before discussing the actual system, let's review a little about headphones themselves. While there are many makes and styles, most modern studio headphones fall into one of two categories: low-impedance headphones, typically 4 to 50 ohms, and high-impedance 'phones, typically 600 to 2,000 ohms. It is not my intention to discuss which of the two categories is better (they both have advantages and disadvantages), but rather to suggest ways to get the best performance from the category you choose.

Let's first talk about the headphone itself. Headphones convert electrical energy into sound waves that can be measured in terms of a sound pressure level (SPL). Assuming we have a flat responding headphone over the audio band, the SPL that the 'phone puts out is determined by the electrical power put into it. In actuality, however, there is an efficiency factor involved, but for simplicity we will assume 100% efficiency. The trick here is to deliver enough distortion-free power to the 'phones to obtain the desired SPL. If you overpower them you risk destroying the transducers (and possibly someone's ears), but if you underpower the units you might not be able to hear them. And if you try to push the driving amp to its limits in an attempt to get more volume, and instead it clips, that can also destroy your headphones.

So how much power do you need? This depends on the efficiency of the headphones, and the amount of headroom that is necessary to prevent clipping. Typical average power listening levels for headphones are only in the tens of milliwatts. Music, however, is very complex in nature, and has many transient peaks. So, even though the average power levels are low, we must use a driver with a power rating that accounts for these peaks. I generally design systems that can deliver between 0.1 and 1 watt of average power to each 'phone. In this way you allow for peak power levels (headroom), and have margin left over to account for variances in headphone efficiencies.

To better understand how electrical power is delivered to a headphone, Figure 1 shows a basic equivalent circuit of an amplifier and a headphone. Assume for simplicity that each transducer can be considered a purely resistive load of impedance R_L. The amplifier is modelled as a voltage generator with a value of V_G volts RMS, and a series internal resistance, $R_G=0$, so that $V_G=V_L$. Remembering Ohms Law, we can work with the following three equations, where all voltages and currents are RMS values:

1) $V_L/R_L=I_L$
2) $P_{ave}=(V_L)/R_L$
3) $P_{ave}=I \times R_L$

Equations 2 and 3 give us average power in terms of R_L (the headphones). So how do we get a lot of power to R_L? Equation 2 shows that if V_G is a fixed value, you can decrease R_L to obtain more power into the 'phones, but equation 1 shows that this is achieved only at the expense of more current; such is the case for low-impedance 'phones. To obtain ample low-impedance power, you don't need a lot of voltage swing, but you do need more *current.* Conversely, for high-impedance 'phones the opposite is true. As an example, let's make $R_L=4$ ohms (low-Z). Equation 2 tells us that to obtain the same power for $R_L=400$ ohms we must increase $V_G=V_L$ by a factor of 10! But equation 1 shows that the current will now decrease by that same factor of 10, so that power is conserved (ain't nature wonderful)!

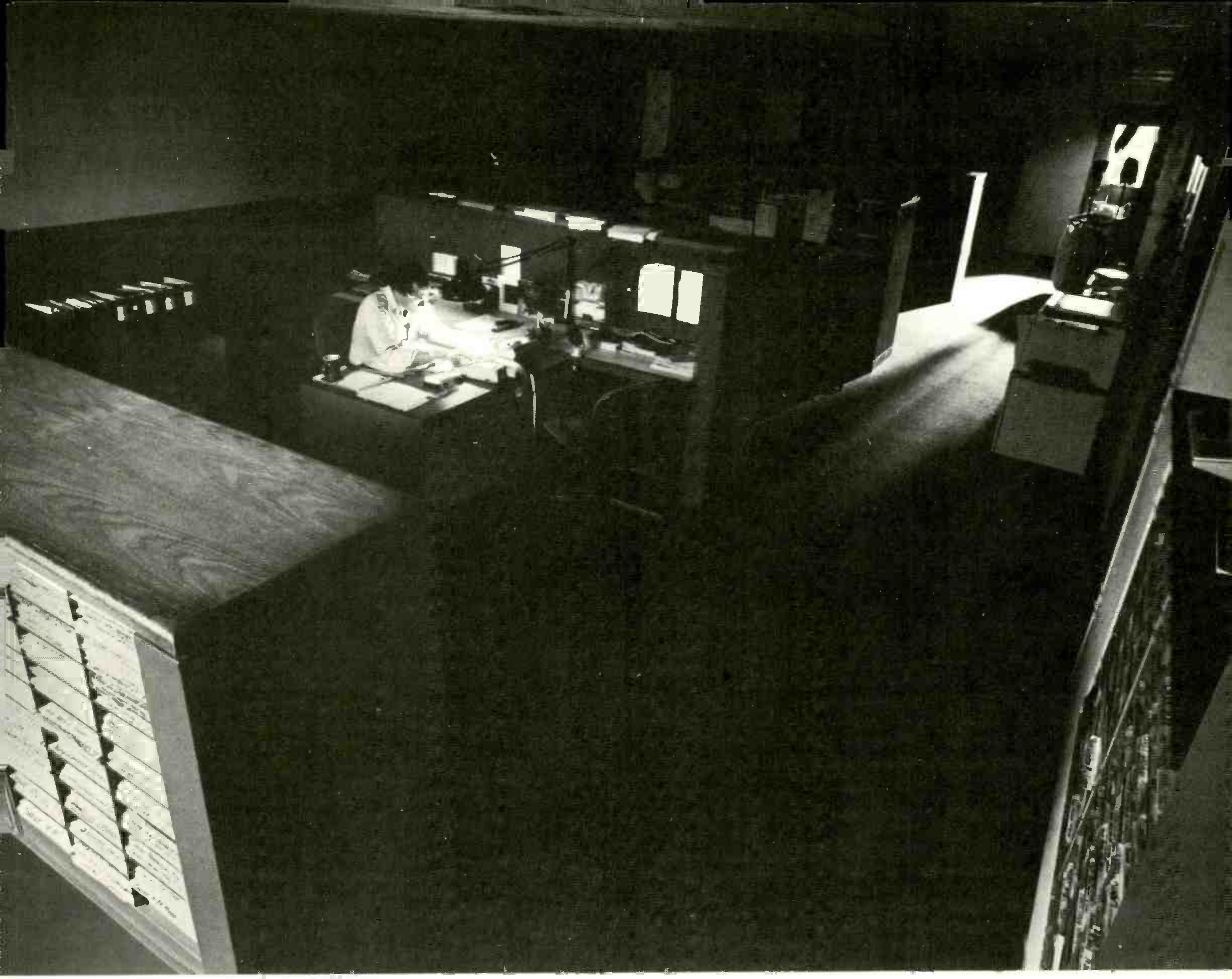
Okay, so what have we said so far? Basically that high-impedance headphones need voltage headroom and not a lot of current, and low-Z 'phones need current headroom, but not a lot of voltage. Now comes an important point of confusion. The model of the amplifier in Figure 1 with $R_G=0$ is, in most cases, very applicable, since most modern amps have very low output impedances. But if we want to maximize our power delivered to RL, shouldn't we make $R_G=8$ ohms? The answer is *No!*

Now before you go running to your electronic books, here's why. Whenever you have a given generator with a fixed and non-changeable output impedance, and you want to deliver the maximum power to the load, you make $R_L=R_G$. If, however, you have a fixed non-changeable load, and you can control the generator resistance, making $R_G=0$ is where maximum power transfer to R_L occurs. To prove this, let's assume that $V_G=8$ VRMS and $R_L=8$ ohms. If $R_G=0$, then $V_L=V_G$ and, by equation 2, $P_{ave}=(V_L)/R_L=8$ watts. Now if we have a voltage divider and $V_L=V_G/2=4$ VRMS. Using equation 2 again, we find $P_{ave}=(V_L)/R_L=2$ watts. Which is why power amplifier manufacturers make their output impedances as low as possible.

Now let's take a look at an ideal distribution system for driving any impedance headphone. Figure 2 shows the block diagram. In this set up, a stereo line-level cue send from the console is fed through low-power audio taper pots (they can be one stereo or independent left and right controls), and then fed to a stereo power amp. While this is the most efficient and ideal approach, it is also the most expensive. The cost of separate power amps, power supply, volume controls etc, can start to add up if you wish to have separate control over many 'phones. It is very feasible, however, to build this exact system at a fairly reasonable price for low-Z headphones, especially with power amp modules that have a complete stereo PA in one package.

Even though the system shown in Figure 2 is ideal for both high- and low-impedance 'phones, it is really only practical for low-Z. Let's look at another example to see why. Assume we want to build a system that can deliver 1 watt maximum power to the headphones. For 8-ohm 'phones, using equations 1 and 2 we find we need 2.83 VRMS and 0.35 amps RMS.

Remembering that the peak-to-peak voltage, $V_{p-p}=V_{rms} \times 2 \times \sqrt{2}$, we see that we must provide 8 volts of peak-to-peak voltage swing. This system could be built using a +5 volt regulated supply, and power amp chip. The power supply could consist of three-terminal regula-



tors, and there are lots of PA chips to choose from. All of these parts are relatively inexpensive and readily available.

Now look at the same situation for 600-ohm 'phones. From equations 1 and 2 for 1 watt, Vrms=24.5 volts, Irms=0.04 amps, and using the equation for peak-to-peak voltage, Vp-p=69.28 volts. To really build the set-up as shown in figure 2, you would need a +35 volt power supply, and a power amp capable of swinging that much voltage, even though it need only source 0.04 amps. Although one-channel, high-voltage PA are available in one package, they are still expensive, and not as readily available as their low-voltage counterparts. There is a more practical solution for high-impedance headphones.

Some of you may now see why many studios use high-power stereo amplifiers for headphone distribution drivers. You might ask, why do you need a 200-watt power amp to drive headphones, when you only use a few watts to drive all of the headphones in the studio? The reason is that to get just those few watts into high-impedance 'phones you need a high-voltage swing. Remember that the 200-watt rating is into *8* ohms *not* 600. Let's use equation 2 again. From the mathematics we can see that 200 watts into 8 ohms corresponds to 40 VRMS. If we now plug back into equation 2 with V_L=40 VRMS, and R_L=600 ohms, we find that this is only 2.67 watts! That big fat 200 watt amp can only put 2.67 watts into 600 ohms. (It can only put 0.8 watts into 2,000 ohms!) Now we see why it is not absurd to use a high-power amp for high-impedance headphones, since we need the voltage swing. Again, you could build your own high-voltage, low-power amp for hi-Z 'phones, but they are not as common or inexpensive to buy.

So we can now understand the more practical solution for hi-Z 'phones shown in Figure 3. Even though this circuit is not the ideal system shown in Figure 2, if it is built properly there are no real compromises. In this set up there is only one power amp that generally can handle more headphones than you could ever need. Note that the distribution system is passive which, unlike the low-Z active system, requires no AC power. Long headphone cable runs are not a concern in high-Z set ups, because the low current draw of high-impedance units makes them much less immune to cable IR voltage drops with which low-Z 'phones have to deal.

The critical area in this system is the choice of pots — it is important to use a high-quality, low-impedance (usually 1 kohm or less) audio taper power pot, and not to exceed its power rating. Also stay away from wirewound pots, as they can make unwanted scratchy noises. I highly recommend conductive plastic power pots like the ones made by Clarostat Manufacturing, Inc. One in particular that I often use is a dual audio taper, two-watt, conductive plastic, sealed 1 kohm pot. It is very high quality, and the two-watt rating makes it capable of handling 44.72 VRMS (which is the RMS voltage put out by a 250-watt/8-ohm amp). The 100-ohm, one-watt series resistors serve three purposes:

1). To provide electrical isolation between outputs regardless of pot position;

2) To protect the power amp against shorts when plugging 'phones in and out of connectors'; and

3) To serve as a headphone "in-series" protection device.

You may wonder why there are not many passive distribution boxes made for low-Z headphones. The reason is that you would need very low impedance but very high power pots. As an example, with a 50-watt amp and an 8-ohm pot, the latter needs a 50-watt rating. And if you could get one of good quality, you may find that it costs you as much as the amp!

There is a lot more that could be said on this subject, but by now we should have a good enough understanding to make some rules of thumb to remember in designing your distribution system. These can be summarized as follows:

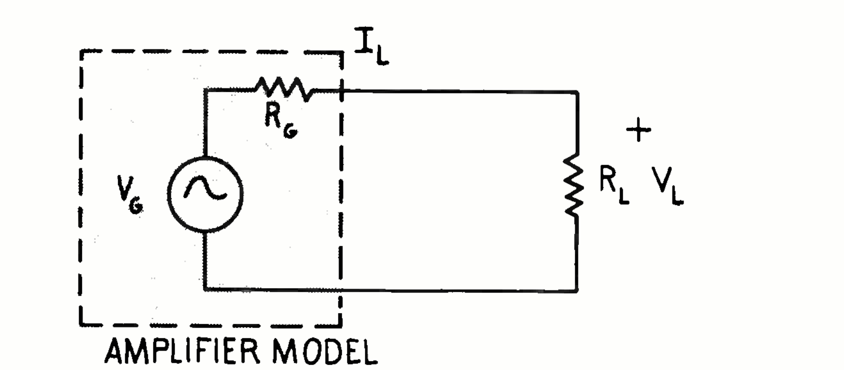


Figure1: Basic Equivalent Circuit of Amplifier and Headphone.

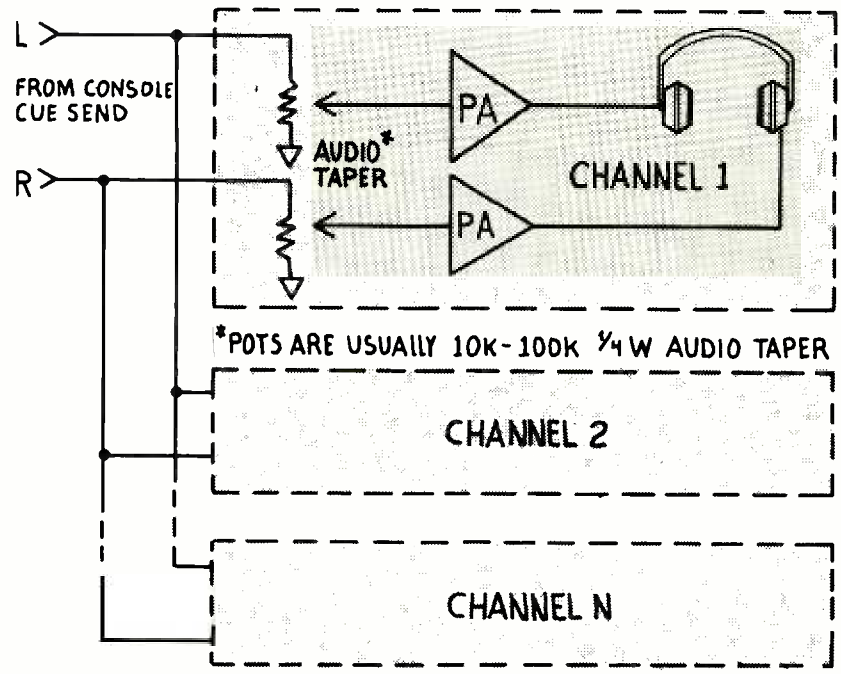


Figure 2: Block Diagram of Headphone Distribution System.

Low-Impedance Headphone Distribution Systems

1). Use an active distribution system that has line-level inputs and individual driving amps, as shown in Figure 2. Find out what levels of voltage and current it can put out with all channels loaded to see if it will meet your power needs. Just because it is active doesn't necessarily mean it was built correctly.

2). If you want to install multiple distribution boxes at various points in the studio, run balanced cable from the console cue-send output, and put access jacks at the desired locations. Make sure there is an AC outlet nearby so you have both power and signal. For cable runs of 20 feet or more it is important for low-Z active systems to use a balanced set up on both driver and receiver ends of line-

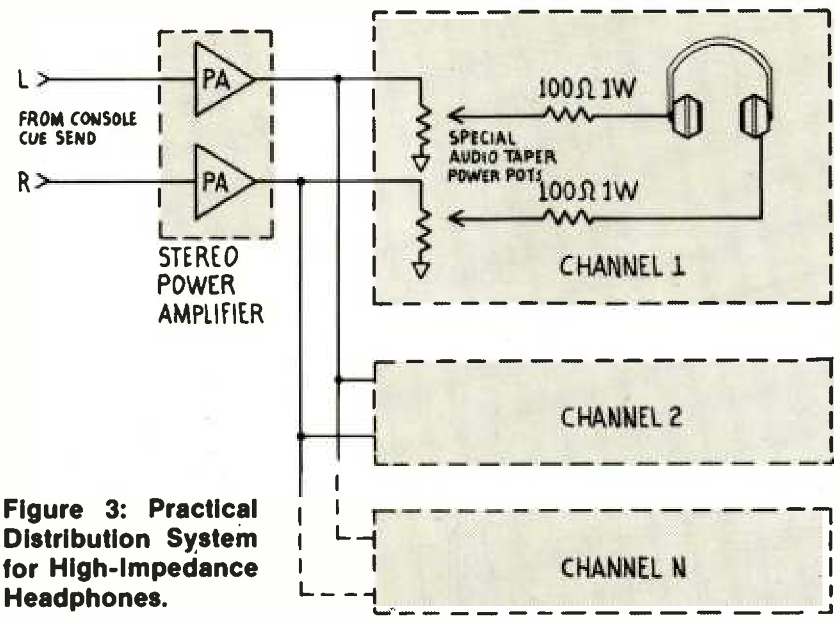


Figure 3: Practical Distribution System for High-Impedance Headphones.

level signals.

3). Passive distribution systems for low-Z 'phones are not practical, and degrade sound quality.

4). Avoid long headphone cable runs after the headphone outputs. With low-Z units the headphone output cable impedance can become a factor. Also, don't try to run several headphones from one output. (Usually two or more is too many.) If you want more outputs than the box allows, get another active box and parallel the line inputs.

High-Impedance Headphone Distribution Systems

1). Use a passive distribution system that has inputs from a stereo power amp. Use a PA that has a minimum of 50-watts/8 ohm-per-channel rating, although you shouldn't need more than 250/8-ohm-per-channel for the most extravagent distribution system. Any amp with a watt/ohm rating inbetween these values should be fine.

2). Generally, you can use multiple distribution boxes on a passive system without a shielded or balanced cable configuration, because of the amplifier's extremely low output impedance.

3). The power pots are the critical component in a passive distribution system; make sure you are using top quality units as previously suggested. If you buy or build the system, make sure you know the type of pots you are getting, their power rating and impedance, what they are made of (stay with conductive plastic), and select audio taper pots of 1 kohm or less. Use equation 2 to calculate power dissapation.

4). Active distribution boxes that were designed for low-Z 'phones can sometimes be used with hi-Z models, except that in most cases the voltage swing on the active box is too low to obtain ample power into hi-Z headphones. This results in not obtaining sufficient volume, no matter how far you turn the control.

5). With most passive boxes, amplifier loading is not a problem. In most cases you can put at least 10 typical boxes with headphones on a single stereo power amp without any loading effects.

Final Note: Headphones of medium impedances — 100 to 600 ohms — can generally be used on either a passive or active system with good results. ■■■

EQUIPMENT ASSESSMENT

APHEX COMPELLOR STEREO COMPRESSOR-LIMITER

Reviewed by Peter Butt

The Aphex Compellor™ is the latest product from the people who gave us Aural Excitement, and is a novel product in a number of ways. Inspection of the Compellor simplified block shown in Figure 1 should serve adequately as proof of that statement. Which would need to be the case if a new program level control device is to survive in a world where limiters, compressors, and leveling devices of all stripe abound, to the confusion of the user. This device has something new to offer in several ways.

The Compellor LED metering displays show considerable thought and ingenuity; they incorporate two-color LEDs in such a way that enables each display function to indicate more than one parameter simultaneously. Input or output levels can be selected for indication by use of a front panel switch. Average level is indicated by red illuminating LEDs, while the peak level in excess of the average is indicated by green. The result is that the approximate average, peak, and peak-to-average ratio of the input or output signal for each channel may be monitored by a single display. Level increments of 3 dB are displayed over a range of +12 to -15.

The LED displays are switchable to indicate gain reduction in a way that shows the combination of leveling and compression simultaneously. The green gain reduction display features a red-indicating LED that separates the two gain reduction levels, showing the leveling function to the left of the red indicator, and the compression function to its right. Peak limiting is indicated by a tell-tale LED for each processing channel.

The unit should be easy to operate — front-panel controls are INPUT gain, OUTPUT gain, and PROCESSING, each separate for each of the two signal channels. Silence sense THRESHOLD, a bypass switch, LED indicator functions, STEREO ENHANCE, and the POWER switch are common to both audio channels. The degree of what is termed "processing" of the audio signal is determined by the setting of the front-panel PROCESSING control. In the extreme clockwise PROCESSING position, the function of the Compellor is to act as a compressor, while in the full counter-clockwise PROCESSING position, the channel becomes a leveling amplifier. Peak limiting function remains operative at about 12 dB above nominal input level without regard to the position of the PROCESSING control.

Another innovative feature of the Compellor is its STEREO ENHANCE mode, activation of which causes the processing function of each channel to be modified by the other, so that the resulting stereo image is one that has a "subtle, natural widening of the stereo image that is fully mono compatible." (This effect has not been experimented with, however, because of time considerations.)

The SILENCE GATE THRESHOLD control allows the user to determine the input level that the device will consider to be silent program, and then hold its side chain processing level at the last active value. This feature prevents the Compellor from mistaking residual noise for low program, and attempting to raise the noise to the desired signal average during pauses in input signal. The processing level will hold for periods exceeding three hours, given that the performance of the sample unit is fairly typical.

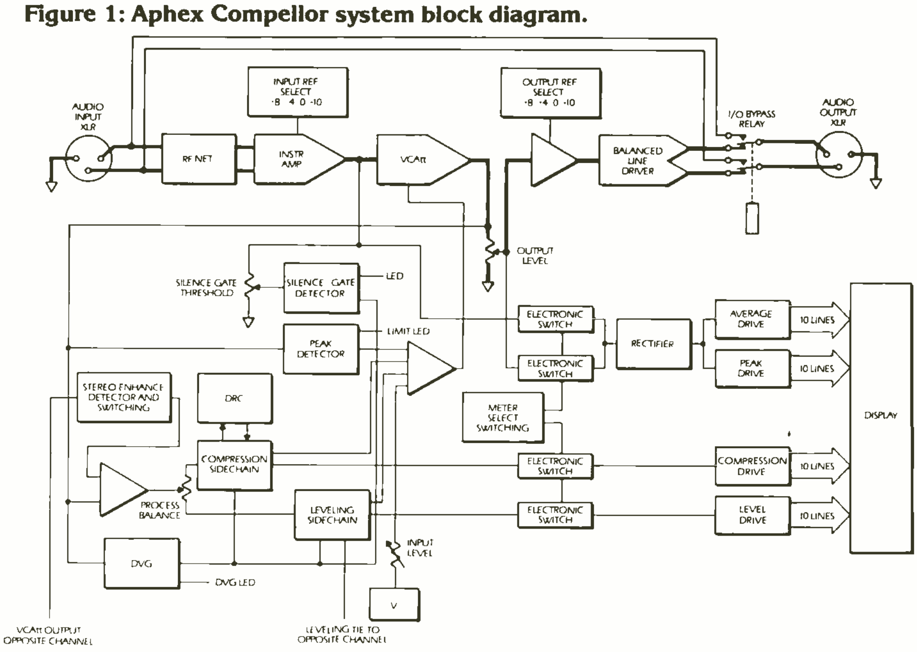
The device may be bypassed by deactivation of a mechanical relay, connecting the input to the output ports of each channel.

Bench Test Analysis

A serious drawback of the commonly-used single op-amp differential input circuit is the fact that the inverting and non-inverting inputs of any real gain block are not as equal as they might seem on first examination. If a driving signal is applied to the inverting input, the signal source impedance becomes part of the op-amp negative feedback circuit. Therefore, any complexities in that source impedance, such as DC-blocking capacitors, become part of the feedback loop character if the many system interface problems whose symptoms seem to defy Logic and Nature. Aphex has chosen to dodge this problem by presenting a non-inverting gain block input to the HI and LO sides of the input ports. The input circuit feedback loop can then be contained entirely within the Compellor where it can be managed within known bounds.

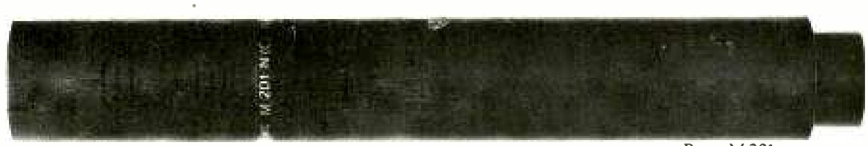
The first technical design feature to strike this reviewer's notice was the Compellor's novel input and output interface circuitry. As can be seen from Figure 1, the input port "looks" into a passive ferrite-core device that is intended to passively eliminate RF signals of a common or normal-mode nature at the input. The input is actively balanced, with DC decoupling provided. Common-mode rejection is enhanced by feedback of the vector difference of the differential input signal to a resistive network at the input amp ports. This permits active cancellation of low-frequency common-mode signals to an exceptional degree. . . *continued overleaf —*

Figure 1: Aphex Compellor system block diagram.

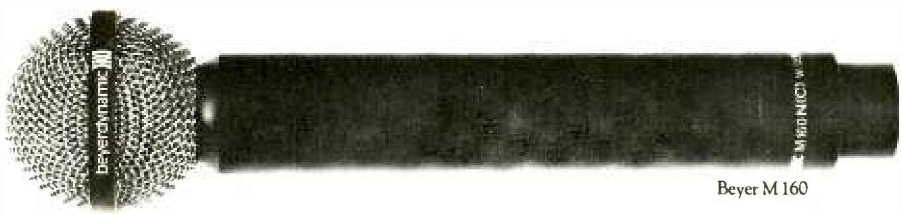


Why Beyer microphones give you more extraordinary performance for the most ordinary applications.

Beyer M 88



Beyer M 201



Beyer M 160

There are other microphone alternatives when high sound pressure is a factor.

As Sennheiser claims, the MD 421 undoubtedly stands up to extremely high decibel levels and has other features that have contributed to its popularity. But if you're already using the MD 421 to mike loud instruments or voices, we suggest that you investigate the Beyer M 88.

The Beyer Dynamic M 88's frequency response (30 to 20,000 Hz) enhances your ability to capture the true personality (including exaggerated transients) of bass drums, amplified instruments and self-indulgent lead vocalists.

The Beyer M 88 features a matte black, chromium-plated brass case for the ultimate in structural integrity. Beyer microphones are designed for specific recording and sound reinforcement applications.

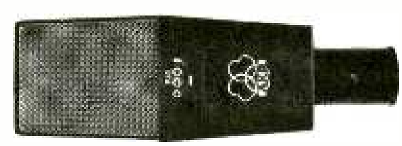
When you need a rugged and versatile microphone, consider the alternatives.

For over 10 years, engineers have used mics like Shure's SM57 for the widest variety of applications in the studio. And we feel that one of the main reasons more engineers don't use the Beyer M 201 in this context is simply because they don't know about it. Those who have tried it in the full gamut of recording situations have discovered how it can distinguish itself when miking anything from vocals to acoustic guitar to tom toms.

The M 201's Hyper-Cardioid pattern means that you get focussed, accurate reproduction. Its wide and smooth frequency response (40 to 18,000 Hz) provides excellent definition for the greatest number of possible recording and sound reinforcement situations.

Each Beyer Dynamic microphone has its own custom-designed element to optimize the mic's performance for its intended use.

You may not always need a condenser microphone for "critical" recording applications.



Some engineers prefer condenser microphones like the AKG C 414 to accurately capture the subtle nuances of a violin or acoustic piano. But should you have to deal with the complexity of a condenser system every time this kind of situation comes up?

The Beyer Dynamic M 160 features a double-ribbon element for the unique transparency of sound image that ribbon mics are known for. While its performance is comparable to the finest condenser microphones, the M 160's compact size and ingenious design offers significant practical advantages for critical applications.

Beyer Dynamic microphones offer state-of-the-design technology and precision German craftsmanship for the full spectrum of recording and sound reinforcement applications.

The Dynamic Decision **beyerdynamic))))**

*Documentation supporting specific comparative claims available upon request. Beyer Dynamic, Inc. 5-05 Burns Avenue, Hicksville, New York 11801 (516) 935-8000

For additional information circle #106

How exceptional is illustrated by the spectrum analyzer 'scope photo of Figure 2. The scale factors are 10 dB per vertical division, while the horizonal scale is log frequency from 20 Hz to 43 kHz; the top trace is the tracking generator output at a -10 dBv level fed to the HI and LO signal inputs, tied together and referred to signal ground; and the lower trace is the signal appearing at the output of the Compellor with the device previously set to unity gain at 1 kHz, +4 dBv input. Common-mode rejection exceeds 60 dB for all frequencies to the left of the 10 kHz graticule line, and never gets near the claimed 40 dB at any frequency below the 43 kHz extreme of the sweep. Not at all bad.

One reason for this fine performance is shown in the input impedance data of Figure 3. There are three traces shown, providing device input impedance for three combinations of drive connection: pin #3 HI, pin #2 LO and grounded to pin #1 (normal mode); pin #3 open, pin #2 driven referred to pin #1, pin #2 open; and pin #3 driven referred to pin #1. The unbalanced impedance holds constant at about 21 kohms below about 2 kHz, where it starts to rise to about 35 kohms at 50 kHz. The curves showing the magnitude of the impedance from either side of the input signal line to ground, labeled 1-2 and 1-3 in Figure 3, substantially overlay one another below about 10 kHz, as confirmed by Figue 2. Agreement between them, and therefore the degree of input circuit balance, is preserved exceeding the 1% imbalance (40 dB CMRR) claimed. Impedance phase curves have been overlaid here as they show the resistive nature of the input impedances below about 10 kHz, becoming somewhat inductive due to the input low-pass filter at higher frequencies. This is one of the best input circuits I have seen to date.

Similarly, the output impedance data shown in Figure 4 shows an output-port unbalanced impedance of about 28 ohms below 20 kHz, while the impedances from each side of the output line to ground are about 120 ohms. Typically, this capacitively-coupled output circuit source impedance shows capacitive tendencies at extremes of frequency, being substantially resistive through the mid-band frequencies. A bridging-load impedance is indicated as optimal, although the device performs creditably driving a 600-ohm resistor. Also, these magnitude and phase curves substantially overlay one another. Thus, we may be assured that in no way will this device upset the balanced condition of any system to which it may be introduced.

The variance in these observed impedance magnitudes from the values published by Aphex has given me cause to recheck my figures using "quick-look" sinusoidal methods. Impedance magnitude figures obtained in that way closely approximate the results of the Fourier transform manipulations shown about the mid-band. Figures 3 and 4 are therefore confidently presented as typical of the device presented for examination.

The Compellor has been reputed to have

Figure 2: Logarithmic 20 Hz to 43 kHz sweep showing common mode rejection. Top trace is the -10 dBv input signal, the bottom trace output for the common mode feed. Vertical scale: 10 dB per division.

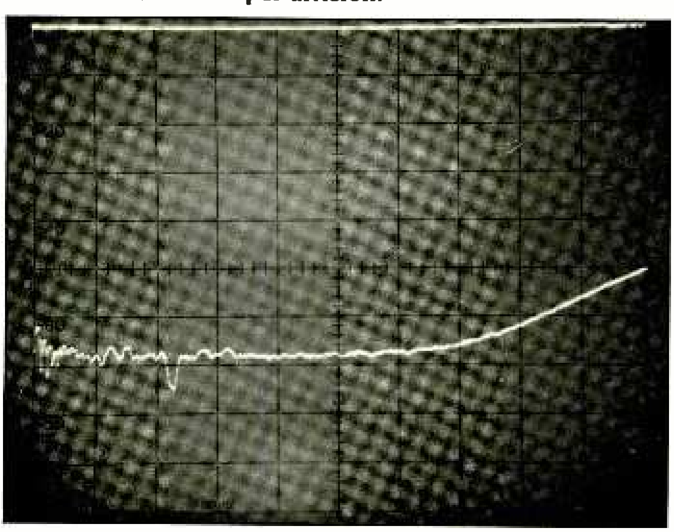


Figure 3: Input impedance versus frequency for three different drive conditions.

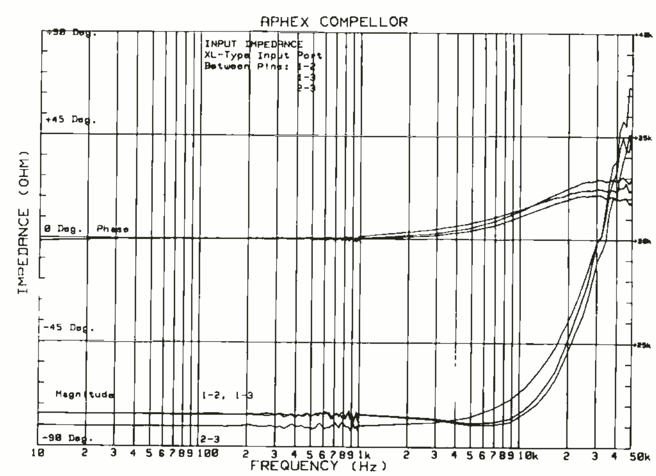


Figure 4: Output impedance versus frequency for three different drive conditions.

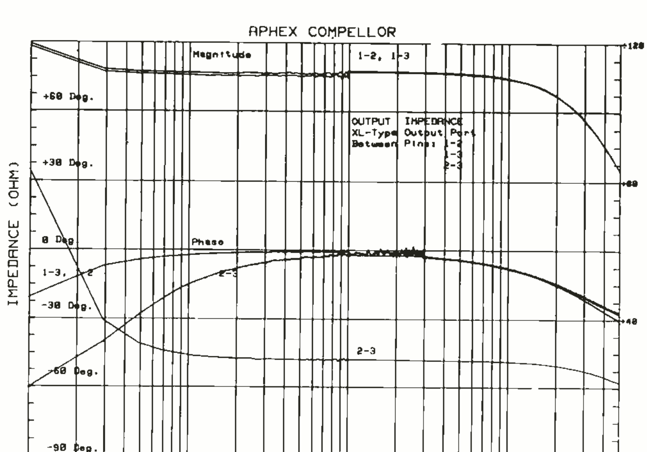
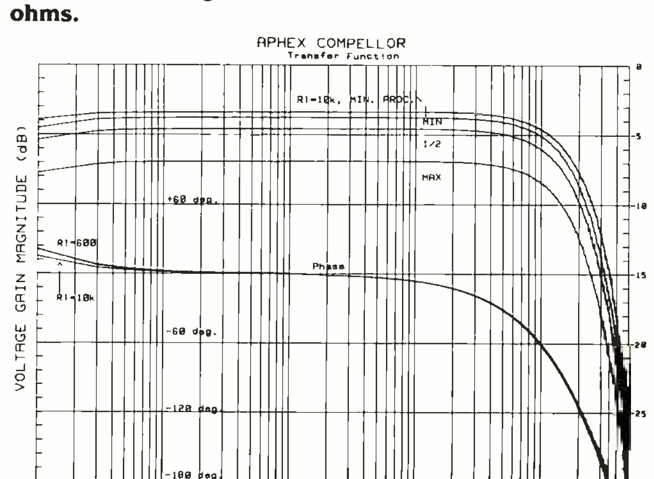


Figure 5: Signal-chain transfer function for three PROCESSING settings; load resistances 10 kohm and 600 ohms.



The surprise is in the price. $425.

The New Orban Model 412A Compressor/Limiter

The new 412A Compressor/Limiter is Orban's entry into the general-purpose level control sweepstakes—it's designed to make you, the audio professional, the winner! Available in mono (412A) and stereo (414A) versions, it's loaded with features that most other low-cost units don't offer—but the real proof is in the *listening.* We invite you to compare its natural, transparent sound to any other compressor/limiter you might know or use. After you do, we feel confident you'll make it *your* essential AGC. Contact your Orban dealer for a demonstration, and find out how affordable Orban-quality processing can be!

Performance Highlights

- ☐ Streamlined, straightforward front panel offers the most-demanded user controls, including ATTACK TIME, RELEASE TIME, RATIO, and THRESHOLD. These wide range controls permit extremely natural sound or special effects.
- ☐ Exclusive Orban feedback control circuitry (adapted from our popular 424A Gated Compressor/Limiter/De-Esser) achieves remarkably transparent sound.
- ☐ User controls interact intelligently to simplify and speed setup, and to prevent errors.
- ☐ Peak limiting and compressor functions are cross-coupled to eliminate potential pumping and modulation effects.
- ☐ THRESHOLD control with 20dB range allows user to determine the level at which gain reduction first occurs, without changing below-threshold gain. Ideal for sound reinforcement applications.
- ☐ Proprietary circuitry achieves optimum headroom and signal-to-noise regardless of THRESHOLD control setting.
- ☐ Front-panel OUTPUT ATTENUATOR control with OUTPUT CLIP LED to indicate line amplifier clipping.
- ☐ Illuminated, true peak-reading GAIN REDUCTION meter is more accurate and readable than LED displays.
- ☐ GAIN REDUCTION OVERLOAD lamp warns of control circuit overload due to a demand for G/R which exceeds the range of the VCA.
- ☐ Hard-wired system bypass switch for fail-safe protection.
- ☐ Side-chain externally accessible for special effects such as frequency-selective limiting.
- ☐ Active-balanced, floating input and output interfaces easily to any system, balanced or unbalanced.
- ☐ Proprietary Class-A Orban VCA features very low distortion and noise.
- ☐ Stereo 414A has STEREO COUPLING switch to permit either stereo or dual-mono operation; an unlimited number of units can be wire-coupled to track 0.5dB.
- ☐ All-metal chassis with RFI suppression on input, output, and AC leads.

Suggested List Prices:
Model 412A (Mono), $425.
Model 414A (Dual Channel/Stereo), $799.

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a minimal audible effect on the signals passing through it, regardless of control settings or the character of the signal. The implication of this is given in the data presented in Figures 4, 5, 6, and 7.

Figure 5 shows the transfer function data for the Compellor for a CW squarewave input signal condition. The time-domain data for the input reference signal and for the various output signal conditions was Fourier-transformed and manipulated to yield the magnitude and phase data shown in the frequency domain. The top-most curve shows system response for the condition of a 10 kohm output load with the Compellor PROCESSING control at full counter-clockwise (*min*). The trace immediately below it shows the function for the identical case with the Compellor output loaded by a 600-ohm resistor, as a worst-case situation. Except for less than a dB of level difference, they are substantially identical. The same can be said for the other two magnitude response curves for the conditions of the PROCESSING control at mid (½) and full clockwise (*max*) settings; they all nest nicely. The Compellor transfer function phase data for all cases overlay exactly, within measurement limits, with the exception of a slightly higher phase lead below about 30 Hz for the case of the 600-ohm output load condition. This is a truly exceptional result for any signal level controlling device.

It is convenient to digress at this point for a brief discussion of the effects of the Compellor PROCESSING control, which allows the user to select a combination of two level control functions to be operative to varying degrees at any one time. The peak-limiting function is internally preset, and is not affected by the PROCESS control setting. The extreme clockwise setting of the control allows compression to dominate the Compellor channel function. Full counter-clockwise setting of the PROCESSING control minimizes compression, and allows the leveling function to dominate. A mid-range setting permits a combination of limiting and compression. There are *no* attack or release or threshold controls available to the user, since these variables are determined by the internal level control systems, and are also signal-dependent.

This fact becomes evident while attempting to arrive at a verifiable value for the device's specified compression and leveling performance characteristics. Changes in signal level, signal burst duty cycle, and duration cause different numbers to be obtained for attack and release times. The observed data given in the Specification Tabulation of Table 1 are thus qualified, as Aphex is not highly specific as to how Compellor attack and release times are arrived at or defined. The criteria given for attack and recovery are the reviewer's own, and were motivated primarily by convenience at the time of observation.

To proceed further, the raw time-domain data that yielded the Compellor transfer function data of Figure 5 have been

Figure 6: Group delay response versus frequency for three PROCESSING settings; load resistance 10 kohm and 600 ohms.

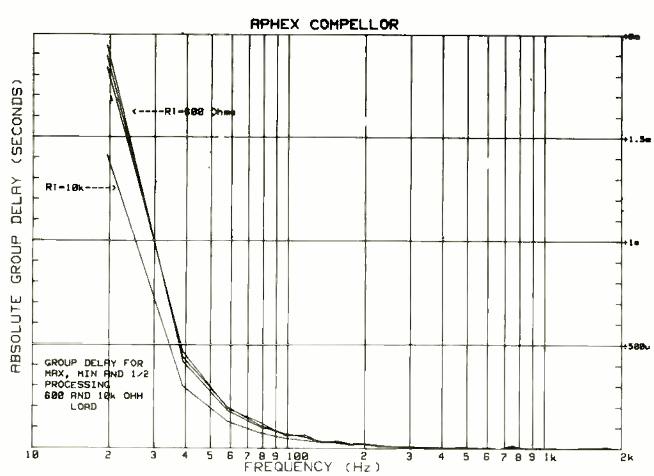


Figure 7: Group delay response versus frequency for three PROCESSING settings; load resistance 10 kohm and 600 ohms.

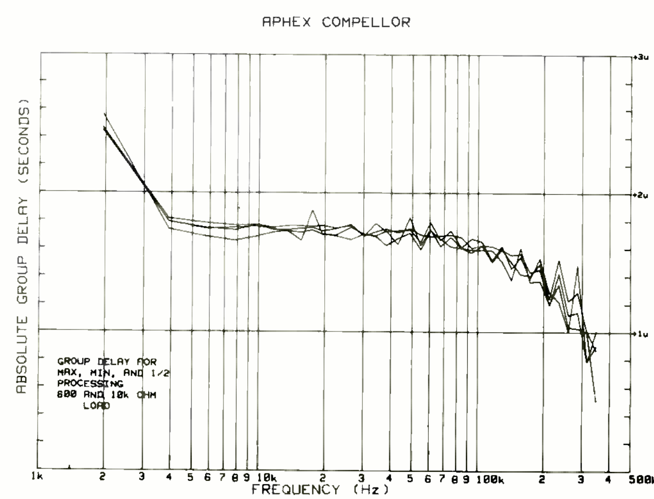


Figure 8: THD products for a 1 kHz drive signal, set for unity gain at +4 dBv. 0.2 mS/div, horizontal scale. PROCESSING: Top trace: max; Center: mid; Bottom: min. Sinewave is output reference.

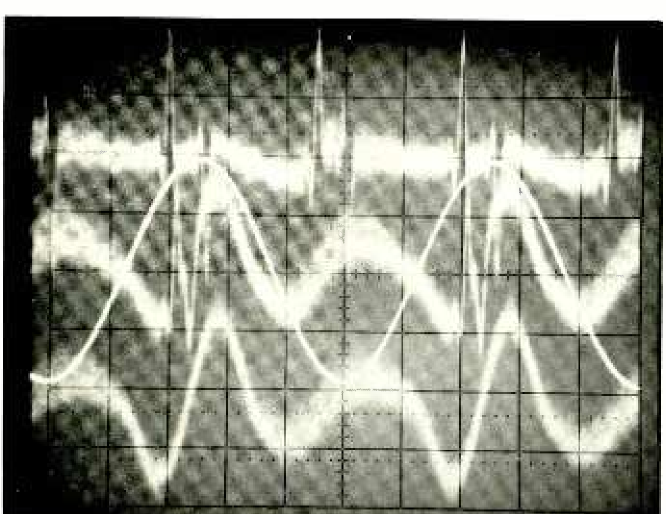
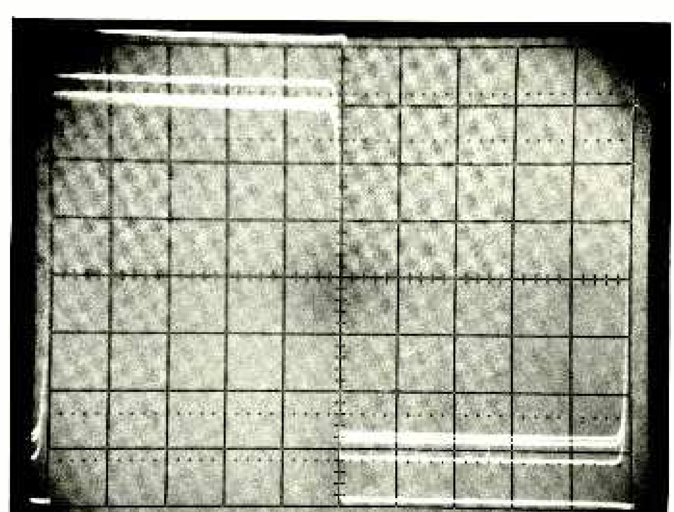


Figure 9: Five kHz squarewave response for PROCESSING settings of max, mid, and min. Top trace: 0.2 V/div; Center: 0.5 V/div. Time base: 20 uS/div. Input reference and outputs shown.



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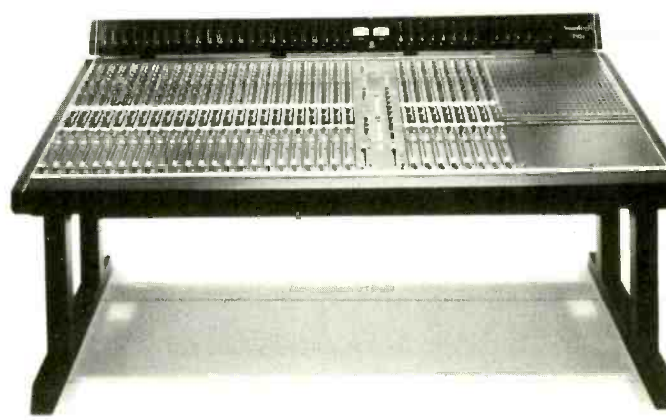

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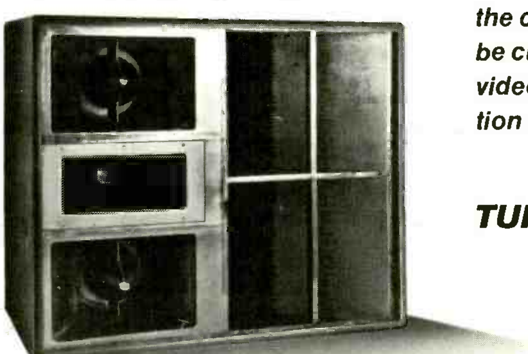

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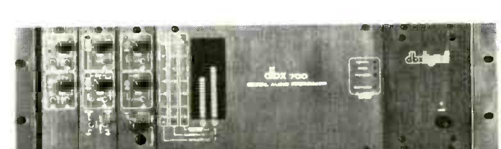

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SAN ANTONIO *HOUSTON* *DALLAS* *MEXICO CITY* manipulated further to yeild the group delay data shown in Figures 6 and 7. Figure 6 covers the band from 20 Hz to 2 kHz, giving an excellent group delay maximum of less than 2 mS for the worst-case of a 600-ohm loading condition. Use of a 10 kohm load brings this figure down to about 1.5 mS. The reader will note how closely the 600-ohm curves overlay one-another for the extreme and mid settings of the PROCESSING control. This is continued in the overlays shown in Figure 7 for the band of 2 kHz to 350 kHz. The four group delay traces are substantially the same, within the limits of a 12-bit digitizer and the noise constraints of the system. It must be remarked that this is a phenomenal performance for a single device operating under conditions of very high signal drive, and varying levels of limiting and compression.

Finally, we come to the traditional THD test results. The oscilloscope photo of Figure 8 shows the reference 1 kHz sinewave superimposed on the distortion products for each of the extreme and mid positions of the Compellor PROCESSING control, under conditions of about 20 dB of gain reduction with the INPUT/OUTPUT controls set for unity gain at +4 dBv, 1 kHz. The top trace reads 0.066% THD for a max setting; the center trace reads 0.062% for a mid setting; and the bottom trace reads 0.032% for a min setting. Obviously, this data was taken using standard CW techniques. The Aphex specification as "Dynamic THD" remains unclear to me.

As far as time domain data are concerned, Figure 9 shows the 5 kHz squarewave response, which is seen to be very clean and free of ringing or overshoot. This photo is a multiple exposure and was taken for the case of max, mid, and min PROCESSING and from a condition of maximum indicated gain reduction. It is notable only for its lack of undesirable artifacts. The largest squarewave is the Compellor input signal. The response for other PROCESSING control settings differ only in amplitude and are overlaid.

Figure 10 shows the effect of the peak limiting function on a 5 kHz squarewave two-cycle burst having a repetition inter-

. . . continued overleaf —

Table 1: Summary of Aphex Compellor Specifications

Description	Quoted	Observed
INPUT:		
Type:	RF-filtered, true-instrumentation differential, balanced.	Self-evident.
Impedance:	50kohms, balanced	21 kohms across HI and LOW input port terminals, 21.6 kohms between audio ground and HI and LO terminals; measured at 1 kHz, un-balanced.
Common Mode Rejection Ratio:	Greater than 40 dB	Greater than 60 dB for frequencies below 10 kHz; greater than 50 dB below 43 kHz.
Nominal Operating Level:	-10, +4, +8 dBv	Yes. Testing conducted at +4; unbalanced.
Maximum Input Level:	+27 dBv	+24 dBv at +4 input/output strapping; unbalanced in/out.
Side Chain:		
Compression:		
Attack time:	5-50 mS	2.5 mS at apparent threshold for 5% reduction in peak envelope of a 10 mS 20 kHz burst, having a 1.7 second repetition interval.
Release Time:	200 mS to 1 second; program-dependent.	470 mS for recovery to 95% of initial peak burst envelope of a 100 mS, 20 kHz burst, 10 dB over apparent threshold.
Threshold:	30 dB below nominal level with INPUT full clockwise.	29 dB below +4 dBV, INPUT full clockwise.
Ratio:	1.1:1 to 20:1; program-dependent.	Signal-dependent. Cannot verify. Observed greater than 10:1.
Leveling:		
Attack time:	2.5 second; program-dependent.	7 mS for a 5% reduction in peak envelope of a 250 mS, 20 kHz burst, 6.5 second repetition cycle.
Release time:	5 second; program-dependent.	2.23 S for recovery to 95% of initial peak 20 kHz burst envelope, 10 dB over apparent threshold.
Rate:	0.5 to 5 dB per second; signal-dependent.	Signal-dependent, cannot verify; 4.5 dB per second observed.
Threshold:	30 dB below nominal level with INPUT full clockwise.	28 dB below +4 dBV, INPUT full clockwise.
Peak Limiter:		
Attack Time:	1 microsecond	1.03 uS, 5 kHz, 2-cycle burst; 50 mS interval.
Release time:	10 mS	3.1 mS, 5 kHz, 2-cycle burst, 2 dB above apparent threshold.
Threshold:	12 dB above nominal level.	12 dB above +4 dBv, PROCESSING full counter-clockwise, gains set for unity at 1 kHz. [See text.]
Output:		
Type:	Electronically-balanced, transformer-less.	Self-evident.
Frequency Response:	±1 dB, 5 Hz to 65 kHz.	±1 dB, 10 Hz to 80 kHz, unbalanced, 10 kohm load.
Noise:	-72 dBv at +4 dB unity gain.	-75.2 dBv and -74.3 dBv, 150-ohm input termination, 20 kHz bandwidth settings for unity gain at +4 dBv input level, 1 kHz; PROCESSING full counter-clockwise.
Source Impedance:	20 ohm, balanced, 10 ohm, un-balanced. (?)	28 ohms, unbalanced; see curves included in review.
Dynamic THD:	0.1% at 20 dB compression, 1 kHz, +4 dBv operating level.	0.066% to 0.032% at 1 kHz, CW signal, 20 dB compression at PROCESSING full clockwise.
Maximum Output:	+27 dBm, balanced; +21 dBv, unbalanced.	+20.8 dBv, 10 kohm load; unbalanced.
Additional:		
Silence Sense Threshold:	Less than -40 dB to 0 dB below nominal input level.	-50 dB to -9 dB, Re: +4 dBv; INPUT full clockwise.
Signal Interface:		3-pin XL-type female input; male, output.
Polarity:		Non-inverting; pin-3 HI (arbitrary).

Size: 1¾ by 19 by 9 inches (H×W×D). *Shipping Weight:* 11 pounds. *Power:* 90-250 VAC, 50/60 Hz, 20W, IEC power cord. *List price:* $1,195.00. *Manufacturer:* Aphex Systems, Ltd., 1334 Saticoy Street, North Hollywood, CA 91605. (818) 765-2212.

val of 10.79 mS. The vertical scale factor is uncalibrated, slightly greater than about 1 volt per division; the timebase is 5 microseconds per division. The 5 kHz burst has been adjusted to the apparent onset of peak limiting, and then increased by an additional 2 dB to achieve the overshoot at the leading edge of the pulse. The PROCESSING control was set to extreme counter-clockwise during this measurement. The intensified markers indicate a time interval of 1.032 uS from the leading-edge plateau level until the overshoot begins to be controlled by the limiting action. Which illustrates the remarkable speed of the Compellor limiting action, as well as its orderly achievement of the signal ceiling level without ringing or other discontinuous behavior of the modified signal waveform. I have not previously seen this kind of action in an audio signal level control device of *any* type.

As far as clipping behavior of the Compellor is concerned, it is commensurate with the quality displayed by the other aspects of the device. Figure 11 shows an input/output comparison of a 1 kHz sinewave at the onset of input clipping input level is +24 dBv, unbalanced. The lower trace shows asymmetrical clipping of the positive peaks, and traces of some other discontinuities about the negative peaks of the output waveform. Output load value is 10 kohms. If the unit input were driven balanced, the input clipping level specified as +27 dBv by Aphex would likely be achieved. Nevertheless, this performance is not at all shabby.

Measurement of the maximum output level presents somewhat of a problem for this device, as it is impossible to override the internal gain control system. Therefore, a gated signal again had to be used to be able to drive the output to clipping. The results of this test are shown in Figure 12. Minimum processing was selected and the OUTPUT control adjusted for full clockwise. A two-cycle, 1 kHz burst having a 1-second repetition interval was fed into the Compellor input, and its level raised until the onset of output clipping could be observed. Input and output terminations were unbalanced, pin #2 grounded. The negative peak clipped first, at a level of about +20.8 dBv into a 10-kohm load, the positive peak clipping with an additional 2 dB of added input level. Since this is apparently the maximum output level that can be achieved with the minimum of PROCESSING and maximum of output gain for this device, this will have to stand as the maximum unbalanced output level figure.

All of the tests described here were done using an unbalanced drive and output condition. This is because it is difficult to find instrumentation capable of providing the quality of signal drive and measurement linearity and accuracy, plus having balanced capability. The tendency of the Compellor to clip asymmetrically under conditions of high signal levels, high gain reduction, and unbalanced driving and loading conditions drive has been observed throughout these tests. It was first encountered during initial digitizations of the 9.765625 Hz squarewave from which data the frequency domain curves below 1 kHz are derived. At high levels of indicated compression, the negative exponential droop of the Compellor output had a different time constant from that of the positive droop. This asymmetry became painfully notable as it affected the Fourier transformations used to minipulate the raw time-domain data. The transformations of M. Fourier are not tolerant of non-linearities, and some rather strange looking transfer functions resulted until the problem was noted and drive levels adjusted to assure linear circuit performance.

For the case of taking signal connector pin #3 as "high," the non-linearity persistently occurs on the negative-half of the output signal swing for a positive-going transition at pin #3. If pin #2 is taken as "high," the result will be the same: the non-linearities will predominate in the negative-half of the signal swing for a positive-going transition at pin #2.

It would seem that the Compellor would realize its maximum performance if driven and loaded as a balanced device. Unbalanced operation, seen here, is entirely acceptable.

Mechanical and Electrical Construction

Mechanically, the Compellor is housed in a steel chassis having a formed steel cover that is retained by four small button-head, Phillips-type sheet-metal screws. The power transformer is a toroidal type incorporating a Faraday shield. AC power is drawn through an integrated IEC receptacle with a Pi-network L-C lowpass filter, mains voltage strapping, and fuse holder. The unit is intended for mounting in standard 19-inch rack cabinets with the signal in/out three-pin connectors appearing on the rear panel of the unit. Separate power supplies are provided for each audio section.

The regulator heat sinks do run uncomfortably warm to the touch. My deep feelings of insecurity move me to suggest that it may be wise to mount the unit in such a way as to permit free air circulation around the right-hand side of the chassis where the supply is located.

Internal circuitry is contained on five circuit board assemblies: power supply, to each input/output boards, and two each level control boards. Interconnections between boards are by 16-pin DIP plugs and ribbon cable. DC power is distributed by push-pin receptacles mating to pins on each circuit board. All ICs are mounted in appropriate sockets. Components are not identified by schematic call-out, and there is no detailed assembly documentation for the Compellor at this time. Documentation supplied with the review unit was preliminary, and therefore did not treat the technical details of the device as completely as might be desired.

The gain control boards, not identical units, are "piggy-backed" on each I/O board with their control shafts extending through the front panel. Removal of the circuit board assemblies entarils removal of the push-on control knobs from their serrated shafts, and removal of the four front panel retaining screws. The piggy-back boards are then removed by discon-

Figure 10: Peak limiter attack time. Vertical: uncal, approx. 1 V/div; Time base: 5 uS/div. Attack time is taken as the interval between the leading edges of the intensified markers: 1.032 uS. [See text.]

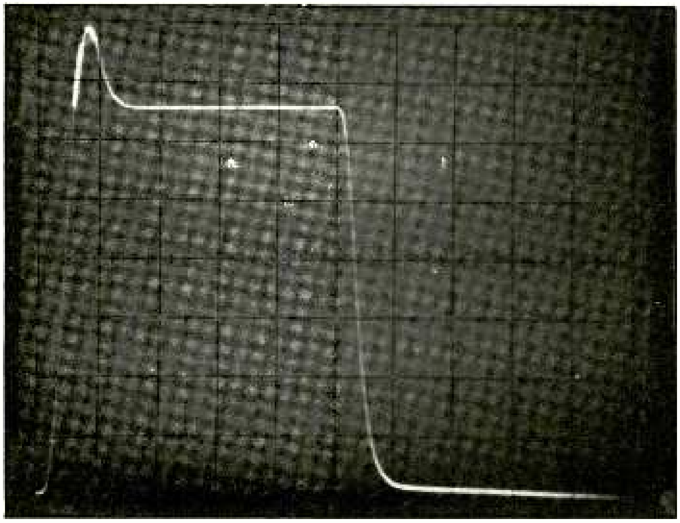
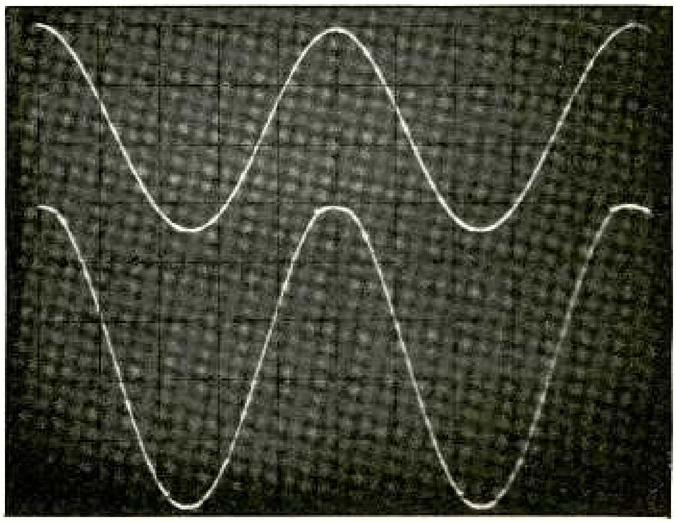


Figure 11: Input clipping characteristics for a 1 kHz, +24 dBv sinewave, Top trace: 10 V/div; bottom: output, 1 V/div, min PROCESSING. Time base: 0.2 mS/div; 10 kohm load.



necting their wire harnesses, and removing the two hex-nuts and two button-head, Phillips-type machine screws. The front-panel control bushing nuts can then be loosened, and the gain control board lifted from the audio I/O board. Optionally, the two hex-nuts may be left untouched, and the gain control board swung up from the I/O board for access to the level gain-strapping and components. If necessary, the I/O board itself can then be removed by taking out two Phillips-type screws and two hex stand-offs. All of this can be done in only a few minutes. The unit may be operated with the circuit boards connected to their harnesses, but removed from the chassis for trouble-shooting.

Only a slight problem was encountered during evaluation of the Compellor. The input stage of channel #2 inexplicably failed near the end of the test sequence. This was largely unnoticed as most of the data shown had been taken for channel #1. A check with Aphex verified that the signal characteristics of that channel are to be expected as typical, and are presented here as such. (They appear to be, as far as can be seen with the unit at hand.) Failure of the common-mode difference amplifier caused a 15 dB shelve-down above about 2 kHz, with other consequences. Replacement of this IC restored most of the performance of channel #2, reducing the shelving effect to about -3 dB above 2 kHz. Isolation and replacement of this device was not particularly difficult, owning to the Compellor's modular internal construction. No further attempt was made to isolate the diminished shelving problem as it did not bear directly on the validity of data already obtained from channel #1.

As for fault isolation, my experience with the adaptive nature of the gain control circuitry performance indicates that simply determining that system performance is within design limits seems to be a formidable task. It would be nice if the final Compellor documentation would provide some procedure for a performance check of the systems. What form such a test would take clearly remains problematic, as considerable ingenuity was required to arrive at the sketchy performance data given above.

I would rate the Compellor maintainability as above average for professional audio gear. This statement assumes, of course, that one can isolate the fault with some certainty. Direct replacement of the individual circuit boards, as described, is a relatively simple task.

Studio Listening Tests

The Compellor was studio tested by Alan Davis, of Total Access Recording, Redondo Beach, California. Davis reports that the Compellor is a device that requires the most studious dedication to precipitate grossly obvious misuse. [In other words, you *really* need to work at it to make it misbehave — *Ed.*] Breathing or pumping could be induced only by setting the PROCESSING control to full clockwise, and the INPUT gain control to maximum. The program material used in the test was rock electric guitar.

The effect of the Compellor was substantially inaudible for all settings of the PROCESSING control. Audible compression effects could be induced only by setting the INPUT to maximum gain and the OUTPUT to minimum gain. Davis found the "breathing" effect thus obtained to be different from results obtained with other limiters, where simple overdriven conditions yeild the commonly-used "wall-of-sound" results so highly-prized by Heavy Metal producers. The results using the Compellor were so transparent as to induce doubt that it was, indeed, working.

Uses for the Compellor might be the preservation of a lead vocal dynamics while retaining a heavily compressed character for the instrumental backing, but still avoiding overloads on the vocal. The term "invisible compression" seemed to describe the Compellor's most noticeable non-attribute.

Davis ventured that the Compellor's price is competitive with other units. The buyer should be aware, however, that he is *not* acquiring a device whose impact on the audio signal can be considered another novel effect.

Aphex promotes the Compellor as a tool for "undetectable compression, increased loudness, and intelligent automatic gain-riding." It would appear that they have succeeded admirably.

The unit's audio path evidently as been carefully designed with attention paid to linearity and gain-stability considerations. The overload characteristics are generally well-behaved, within its operating limits, and do not induce grossly undesirable side-effects such as oscillation or latch-up. The asymmetry of time-domain response noted above for grossly overdriven conditions indicates that the Compellor is not an infallible device and, in fact, can be misused if sufficient dedication is applied. The output evidently can be shorted indefinitely without harm. Balanced or unbalanced operation seems to be irrelevant to the device. Referring to Figures 2 and 3, the term "balanced" is used advisedly here.

The observed port impedances were not close to those specified. Their measured values are entirely usable, and not outside the limits of acceptibility for professional-audio applications. Interface of the Compellor into a large audio processing chain should be a matter of simple procedure. The passive input RF filter and the excellent active low-frequency common mode rejection performance make it hard to go wrong for any application that comes readily to mind. Life's serenity would be greatly enhanced if more equipment manufacturers showed as much concern for the unfriendly environments with which their products must contend. ■■■

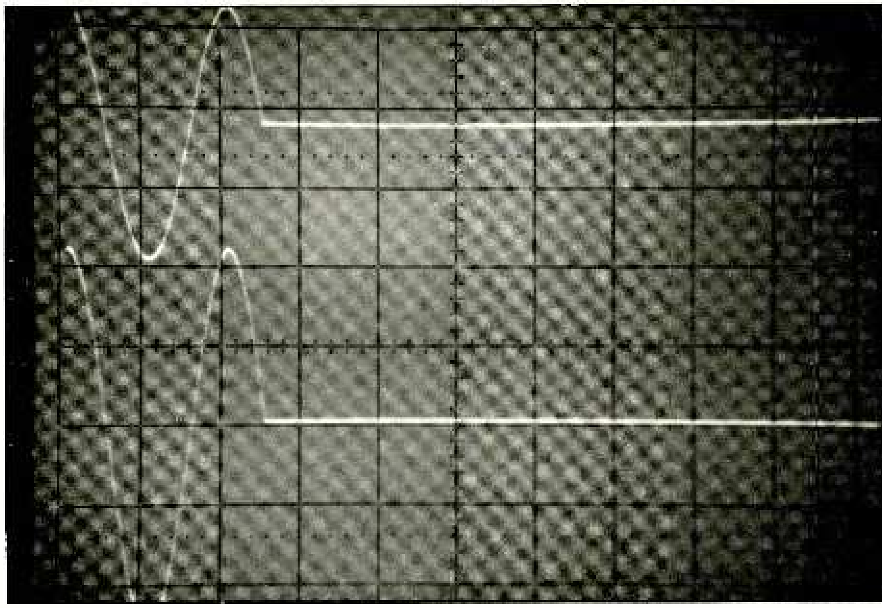


Figure 12: Output clipping. Top: input, 5 V/div; bottom: output, 5 V/div. Time base: 2 mS/div; 10 kohm load [See text.]

New Products

MCI SYNC MASTER SMPTE/EBU SYNCHRONIZER

The Sync Master has been specially designed to contain many features that are not available with any other synchronizer, including up to four tributaries with standard controller; up to a maximum of 32 tributaries with large system controller (to be released); SMPTE/EBU remote I/O standards (studio network); an Edit Decision List of up to 200 edit points with editing facilities; and modular construction with distributed intelligence on the network to assure system speed throughput in all configurations.

Additional features: executive program that monitors activities of all machines in all modes with error reporting capabilities; programs that normally involve changing PROMs on other synchronizers (for example, ADR) are all contained in one unit; complete removable display facilities (Master TC, Slave TC, Offset, Punch-in, Punch-out, Take #) in a small compact unit; and 32-character dot matrix display for EDL prompts, self-diagnostics, instruction, and HELP Menu.

Interfaces are currently available for all MCI machines, Sony BVU-800 and BVH-2000 video transports, plus Studer, Ampex, and Otari machines; others will be made available as required.

Prices range from $12,000 for a two-machine system, to $16,000 for a four-transport configuration.

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The five new amplifiers P-75D, P-150D, P-300D, P-150M, P-300M — are designed for stereo or mono systems; Stereo models ("D") contain a rear panel switch for mono bridging mode.

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Each amplifier's front panel provides green and red LEDs for quick visual confirmation of signal presence, circuit protect/thermal warning, and output clipping.

Rear panels provide XLR and phone jack inputs wired in parallel. Each amp provides an octosocket for an input transformer if balanced input is desired. Stereo models include high-pass filter on/off switches to prevent speaker damage from subsonic frequencies.

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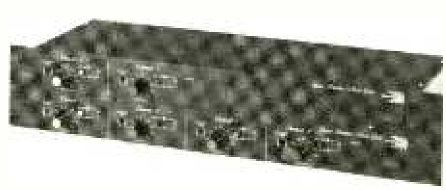
For additional information circle #112

BIAMP COMBINED STEREO LIMITER AND NOISE GATE

The new stereo/quad limiter-compressor and noise gate is designed to handle a wide range of problems associated with limiting overloading from line conditions, provideing speaker protection, and dealing with microphone overload, while the noise gate section can be used to eliminate hum and noise



The two and four independent channels feature front-panel threshold control, plus switchable selection for limiting or noise gate operation. An LED indicates limiting or signal through the noise gate. Release time can be varied by means of screwdriver-adjustable controls on the front panel.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

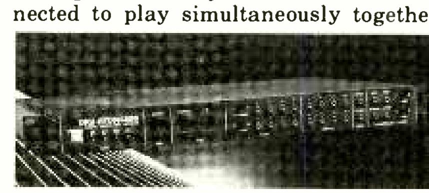
For additional information circle #113

ZYPHER ELECTRONICS NEW DIGI-ATOM 4800 ANALOG-TO-MIDI INTERFACE

The DIGI-ATOM 4800 allows analog sequencers, such as the Roland MC-4, MC-202, and others, to control any MIDI-equipped synthesizer. The unit converts up to eight control voltages and gates into MIDI key data; additional inputs include velocity control, pitch bend, modulation amount, release, and even sound program select. The user can select from one of four different operating modes for eight-voice polyphonic control, or four-voice polyphony with independent velocity control on each voice. Other modes include double mode, and triple mode for a "live" ensemble of MIDI synthesizers.

In studio applications, the unit's double and triple modes can be used to control timbre and articulation as well. By alternating between MIDI synthesizers on different channels, expression, dynamics, articulation, and timbre changes can be performed that otherwise are not possible to play in real time.

The DIGI-ATOM 4800 is also said to offer numerous synchronization and interface applications. A Roland-type Sync Input can be used to synchronize a MIDI sequencer to an analog sequencer, and analog and MIDI synthesizers can be connected to play simultaneously together.



Also, MIDI-equipped drum machines can be driven from analog pulses or gate signals. Even pedals, breath controllers, drum pads, or analog synthesizer CVs can be interfaced to any MIDI synthesizer to control velocity, pitch bend, modulation amount, release, portamento (on Yamaha DX Series synths), or program select.

Export distribution is being handled exclusively by:

TALK STUDIO
1-26-3 ZENPUKUJI
SUGINAMI-KU TOKYO
JAPAN 167
(03) 394-4368

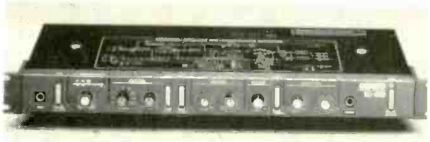
For additional information circle #114

DE-200 BOSS DIGITAL DELAY FROM ROLAND

The rack-mounted DE-200 DDL enables preset amounts of delay to be selected and fine-tuned within a range of 0.5 to 1.0 milliseconds by means of a separate control. The modulation section on the DE-200 features adjustable rate and depth, while feedback level can be adjusted; there is also a Feedback Phase Inversion switch provided.

A novel Rhythm/Sync function enables the DE-200 to be triggered by an external drum machine. The trigger can be used to synchronize delay times to the beat of a song, or syncopated delay rhythms can be created based on the programmed beat. And with the Hold function, any delay pattern can be repeated infinitely for a sound-on-sound type of layered echo effect.

LED displays indicate input gain level for proper adjustment, and a front-panel



output mixer allows the direct and delay signals to be blended in any ratio desired.

Suggested retail price of the Boss DE-200 is $399.

ROLANDCORP US
7200 DOMINION CIRCLE
LOS ANGELES, CA 90040
(213) 685-5141

For additional information circle #115

INTERSONICS MODEL TPL-3 SUBWOOFER

Second generation of the company's servo-drive loudspeakers, the new model TPL-3 incorporates a modular "drive pack" into a hyperbolic horn with over 10 feet of path length for extended response of low frequencies. The subwoofer is configured in a truck-pack cabinet measuring 45 by 45 by 22½ inches.

Since the unit has an extremely high source strength, it means that low bass can be heard and felt at distances where ordinary subwoofers fall off, the company claims. The result is a very efficient sub-bass system for use in a variety of situations, from large-scale sound reinforcement to theatre applications.

INTERSONICS, INC.
425 HUEHL ROAD UNIT 11A
NORTHBROOK, IL 60062
(312) 272-1772

For additional information circle #116

HARRISON UNVEILS "THE RAVEN" MULTITRACK CONSOLE

According to company president David Harrison, "The Raven meets the needs of a special segment of our market which requires a competitively-priced, yet highly responsive console. However, The Raven will retain our 'no compromise' signal handling performance found in our more expensive consoles."

The Raven is offered in a single configuration consisting of a 40-position mainframe that will accommodate 36 input modules. All consoles are supplied with 28 input modules, and three master modules, plus blank panels for the fourth master module position and all eight unused input module positions.

"For customers who desire 32 or 36 input console versions, a console expansion kit consisting of four input modules, a four input module motherboard section, and a power cable is available as a separate expansion kit," Harrison continued. "This kit requires less than an hour to install into the mainframe of the console."

Much like an MR-4 in function, features, and component design, The Raven incorporates a different look from the standard Harrison console. It is supplied in the standard Harrison dark gray and a very light gray silkscreen.

No internal patchbay is supplied; instead, provisions for a user on dealer supplied external patchbay have been incorporated.

HARRISON SYSTEMS
P.O. BOX 22964
NASHVILLE, TN 37202
(315) 834-1184

For additional information circle #117

THREE NEW DEVELOPMENTS FROM SOLID STATE LOGIC

Scheduled to be unveiled at the Paris AES convention in late March, the new Synchronizer Controller should be of special interest to all facilities operating multiple synchronized machines. Interfacing directly to the studio synchronizer, the new unit provides transparent control of the entire machine system directly to the operator via the SSL Primary Studio Computer keyboard. The Master Transport Selector unit enables any of the machines to be automatically selected and controlled as the Master.

The new SL688V Stereo Mix Matrix has been developed for the SL6000E Series Stereo Video System, which offers three stereo mix busses, designated A, B and C, to ena-

. . . continued overleaf —

New Products

ble creation of separate music, dialog, and effects stereo mixes for post-production in live situations this arrangement provides a variety of easily obtained mix and mix-minus feeds. The Mix Matrix is intended to provide great flexibility for both stereo and mono post-production.

Finally, the new Programmable Equalizer, which can be retro-fitted to existing E-Series systems, has two independent three-band parametric equalizers with variable Q, which can be linked to work as a stereo unit. It includes a programmable panpot for each channel, and has a visual display of the actual control value of every variable element. All movements of controls are remembered as part of the dynamic mixing system, and can be replayed and updated subsequently.

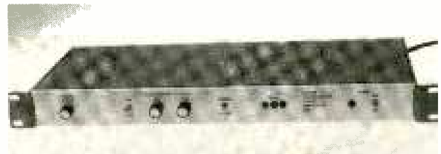
SOLID STATE LOGIC INC.
228 EAST MAIN
MILAN, MI 48160
(303) 439-8516

For additional information circle #119

STUDIO TECHNOLOGIES RELEASE MODEL ANL-1 STEREO SIMULATOR

The new device is designed to create a convincing stereo impression from any mono source. Utilizing new discoveries in how the human binaural hearing system processes information, the Stereo Simulator creates two incoherent signals from one source, similar to those existing at the ears of a listener.

The unit can not only recreate the stereo space and depth but, by using the modulation feature, many new effects are possible, such as stereo chorusing, stereo pitch bends, etc.



The Stereo Simulator is said to be especially useful with purely electronic instruments which are usually assigned to a panpot during mixdown. The device creates a much more pleasing "spread" to the sound, allowing the instrument to be heard even in the most complex of mixes.

Many kinds of stereo simulation are possible using outboard devices such as time delays, pitch shifters, equalizers, etc. Unfortunately most of these attempts are not mono compatible. The Stereo Simulator is fully mono compatible, consequently avoiding these problems.

STUDIO TECHNOLOGIES, INC.
6666 NORTH LINCOLN AVE.
LINCOLNWOOD, IL 60645
(312) 676-9400

For additional information circle #120

BOSE INTRODUCES NEW PROFESSIONAL AUDIO SYSTEM FOR LIVE PERFORMANCE

The new system features 802 Series II Articulated Array loudspeakers, 302 Tandem-Tuned bass systems, and an 802-C System Controller. The 802 Series II speaker, a refinement of the well-known 802, offers increased sensitivity due to use of new 4½-inch D-11B full-range drivers, which feature low-impedance, edgewound aluminum voice coils, 12-ounce Ferrite V ceramic magnets, and weather resistant cones. A refined Directivity Control Circuit is said to improve horizontal dispersion throughout the HF range, and protects the drivers from the effects of HF overload.

The 802-C System Controller combines the functions of three active equalizers, an automatic switching circuit, and an electronic crossover. To select proper crossover functions and equalization curves, the user need only plug into the appropriate rear panel output jacks.

The 302 Tandem-Tuned bass system incorporates a transducer loading system that loads each side of the two 12-inch LF-88-A transducers. The 302 cabinet is constructed of impregnated resin board, and includes recessed handles on the cabinet sides.

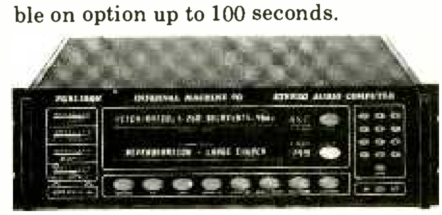
BOSE CORPORATION
THE MOUNTAIN
FRAMINGHAM, MA 01701
(617) 879-7330

For additional information circle #121

PUBLISON LAUNCHES "INFERNAL MACHINE 90" AUDIO COMPUTER SYSTEM

The new stereo-in/stereo- or four-channel output device provides delay from 20 microseconds to 5 minutes, according to selected option, echo with digital feedback, deglitched pitch-shifting, with various algorithms according to input sound (from -2 to +1 octave), automatic arpeggio, reverse, memorised sounds up to 5 minutes (with memory option) various algorithms and frequency-dependent selectable reverberation times, time compression/expansion, and digital editing.

The Infernal Machine 90 has 950 memory positions allowing the non-volatile storage by the user of any function with all parameters, and can be synchronized to SMPTE. Numeric storage of sounds on a Winchester disk drive is also available on option up to 100 seconds.



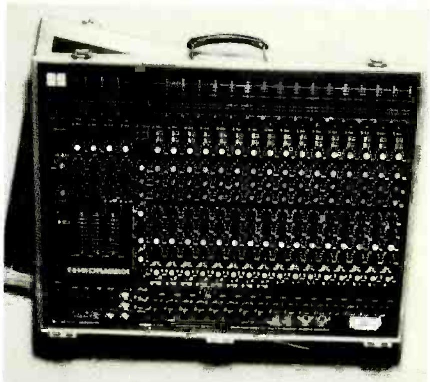
Specifications include 16 bits A/D conversion (19 bits on option), dynamic range of 96 dB for 16-bit option, frequency response 20 Hz to 20 kHz (+0, -3 dB), input level up to +26 dBm, output level +20 dBm on 600-ohm, and sampling frequency 50 kHz or lower on option. Memory capacity is from 5.242 seconds up to 5 minutes. Interface options include SMPTE, MIDI and digital in/out.

PUBLISON AUDIO PROFESSIONAL
RUE CRESPIN-DU-GAST
75011 PARIS, FRANCE
(033) 1-357-6407

For additional information circle #122

SOUNDTRACS S-SERIES OF PA AND RECORDING MIXERS

Semi-modular construction of the 16/4/25 allows easy removal of any input channel. Every channel, including the subgroups, has a pre-EQ inset point of +4



dBm (100-ohm) send, and +4 dB (10 kohm) return levels. The third auxiliary rail can be accessed by the auxiliary #2 return, and wired pre- or post-EQ.

The microphone/line input switch selects either balanced low-impedance or high-impedance inputs. A peak indicator LED illuminates 5 dB before clipping. A 1 kHz oscillator enables line-up of the four output groups, which are monitored by four 10-element LED meters. The three-band equalization encompasses fixed high and low shelving filters, plus a sweepable quasi-parametric midrange of 350 Hz to 8 kHz.

The S Series includes 24- and 8-channel models that have identical features and specifications, save that the multicore connector is absent on the 8/4/2S. All S-Series mixers come complete with heavy-duty flight cases.

MCI, INC.
BOX 8053
WACO, TX 76714
(817) 772-4450

For additional information circle #124

FAIRLIGHT UNVEILS SERIES II X VERSION OF CMI

The Series II X is the expanded memory (500K RAM) version of the CMI. The company's research and development is currently working on the first release of new software that will take advantage of this additional memory capability, to run OS9 software — i.e. "off-the-shelf" accounting, and spreadsheet packages, as well as high-level languages.

The Series II Voice system, a modular hardware option for the CMI, provides 85 dB dynamic range, coupled with a maximum playback frequency bandwidth of 20 Hz to 20 kHz. In addition, its attack transient response is said to be an order of magnitude greater than the standard system.

For the film-music composer, Fairlight recently launched the Film Music Processor, a program designed to act as a combined click track book/stopwatch/scratch pad filing system.

FAIRLIGHT INSTRUMENTS, INC.
2945 WESTWOOD BLVD.
LOS ANGELES, CA 90064
(213) 470-6280

For additional information circle #125

For additional information circle #126

New Products

ADVANTAGE MODEL 310 AUDIO NOISE AND LEVEL METER FROM VALLEY PEOPLE

The Model 310 was created to provide recording studios and broadcast facilities with a low cost, high quality measurement device capable of delivering greater accuracy than that achievable with more expensive general purpose instruments. Careful attention has been paid to ensure that the features are relevant for analyzing noise performance in modern audio equipment.

The unit offers isolated, balanced, Trans-Amp differential inputs to eliminate unwanted noise, RF, and hum pickup, while allowing measurement of extremely low-level signal; 10 Hz to 100 kHz wide band filter that allow insertion of an external weighting network as required be the user's measurement application; 20 Hz to 20 kHz multiple pole filter to achieve accurate measurement of noise in an actual 19,980-Hz bandwidth through incorporation of 18 dB per octave filters with appropriate -3 dB points; 400 Hz to 20 kHz multiple pole filter to eliminate measurement errors caused by ground loops and other low-frequency noise sources; and "A" and CCIR weighting filters; average (VU) RMS, and peak detector response; plus a dual-scale analog meter that displays a wide 70 dB scale in two ranges for measurements from -100 dB to +30 dB.

VALLEY PEOPLE, INC.
2820 ERICA PLACE
NASHVILLE, TN 37204
(615) 383-4737

For additional information circle #128

HAND-HELD CALCULATOR FOR CONVERTING TIME SIGNATURES TO VIDEO AND FILM TIMINGS

The RHODESystems II, a software package incorporated into a hand-held Hewlett-Packard computer, is said to have applications in production, post-production, and music scoring for composers/ arrangers, film/TV editors, producers, and engineers.

The unit can be used to convert any of the following to all of the others: drop-frame SMPTE timecode, non-drop-frame SMPTE timecode, EBU timecode, film frames (any speed), film feet and frames (any size and speed), minutes and seconds, seconds and hundreths. It also adds and subtracts any of these units from any of the others, and emulates the synchronization of film with videotape so that any address in one may be read in the other, and also in elapsed clock time from any start point. (For the rest of the world, the 25 FPS film standard and EBU timecode are included.)

For music scoring, RHODESystems II uses any of these common time units along with standard click track or metronome values to find beat numbers. Knowing any two of the three elements — time, tempo, or beat — the third can be found quickly. Given a range of acceptable click track tempos and the relative importance of each hit in a scene, the computer will figure the optimum click and print a hit list with each hit, and its corresponding beat number, and weight along with the "lag" between the actual hit point and the next whole or half-beat. Since the hit list may be printed using any of the time units, the composer can score using a combination of film and videotape.

RHODESYSTEMS INC.
P.O. BOX 1550
RADIO CITY STATION
NEW YORK, NY 10101
(212) 245-5045

For additional information circle #129

NEW DIGITAL KEYBOARD FROM 360 SYSTEMS

The new unit is described as the first instrument designed specifically to reproduce real acoustic instrument sounds. The string sections, for example, comprise 12 digitally recorded violins, six violas, six celli, and two double bass.

All of the instrument sounds are stored on digital memory chips that are plugged into the instrument by the user. Each instrument is, in effect, custom since the buyer selects only the sounds he wants for his own needs; extra sounds can be added any time.

There were several key design goals that are said to set the Digital Keyboard apart from other instruments. Not only are all the notes real, there are also a large number of "samples" of them across the keyboard. None of the usual odd-sounding side effects show up from stretching one sample over the entire keyboard, the company claims. Each note is the full length normally played on the instrument being copied — many are eight seconds or more in length.

360 SYSTEMS
18730 OXNARD STREET #215
TARZANA, CA 91356
(213) 342-3127

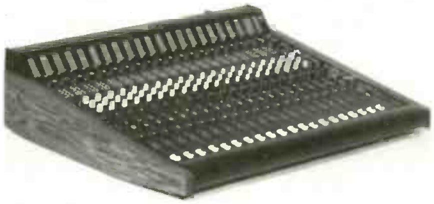
For additional information circle #130

BIAMP LAUNCHES MODULAR BIMIX SERIES OF MIXERS

The bimix Series of modular multitrack recording consoles can be switched into live-performance road mixer by pushing buttons and repatching. The mixer can also be used as a broadcast mixer.

Because of the use of fully modular inputs/outputs, any combination of one to 24 outs and from eight to 40 inputs can be achieved. An input and output mixer and stereo monitor is built into each I/O module, the output section being assigned internally to create the desired configuration.

Mike/line inputs, consisting of four transistors and half a dual 5532-transformerless front end, are said to provide a constant input impedance CMRR and audio bandwidth at any given trim level. (Input transformers are optional.)



The bimix is a full 16-track modular console, with 16 output busses, two cue busses, two effects busses, and stereo monitor. Features include metering that is adjustable to -10 dBV and +4 dB nominal operation; meter select for inputs, outputs, and tape inputs; full patch capability on rear panel; three-band sweep EQ: solo on all inputs and outputs; stereo control room and studio feeds; and a communications module with internal mike, headset connector oscillator, headphone, and external click-track assign control.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

For additional information circle #131

PHILIPS PROFESSIONAL COMPACT DISC SYSTEM

The new system comprises two Compact-Disc drives and controllers. The latter, which is presented in the form of an easy-to-operate, low profile console, has two identical keypads — one for each CD-Drive. In addition, a corporate numeric keypad and a search dial is included. Relevant data for both CD drives is presented simultaneously by video display.

Access to track number, time (from the beginning of the track being played), and time left (remaining time on the track, plus the next pause), can be provided with an accuracy of 13.3 milliseconds. User selectable functions are also available for display, including Absolute Time, Absolute Time Left, ISRC Number of Music, Catalogue Number of Disc, and List (table of contents).

The search dial on the Control Unit enables a fast- or slow-mode search to be carried out through a Disc's information, and the selection of a cue point at any part of the program. A further aid to easier operation is that pre-programming can be undertaken, commands being entered into the system's memory via the control unit.

NEDERLANDSE PHILIPS
BEDRIJVEN, B.V.
THE NETHERLANDS
(040) 7-57279

For additional information circle #132

New Products

NEW SD SERIES LOUDSPEAKER SYSTEMS FROM TOA

The three-way SD speaker system incorporates a moving-coil tweeter, and a high-frequency attenuator for tailoring its output to performance requirements and room acoustics. Other features include bi-amp and bridging connectors, recessed handles; interlocking corners; a stand-mounting adaptor; and a compact enclosure.

The system incorporates Thiele-Small aligned bass reflex design to provide a greater bass range and efficiency and its front panel slotting functions as an acoustic low pass filter.

Overall, the SD Speaker is designed to provide extended frequency response (50 Hz to 20 kHz, Model 38SD), high power capacity (360 watts continuous program), and a very sensitive efficiency rating.

TOA ELECTRONICS, INC.
480 CARLTON COURT
SO. SAN FRANCISCO, CA 94080
(415) 588-2538

For additional information circle #135

E-MU SYSTEMS EMULATOR II DIGITAL SAMPLING KEYBOARD

The Emulator II is described as not just an update of the original Emulator. The new instrument utilizes a new data-encoding technique that is said to result in greatly increased effective frequency response with a decrease in digital distortion. It is available with up to 17 seconds of sampling time, and a full megabyte of disk storage.

The dynamic expressivity of its five-octave, velocity-sensing keyboard makes possible instrumental lines of startling realism. What's more, inclusion of filters, VCAs, envelope generators, and independent delayed LFOs for each of its eight channels allows extensive programmable modification of any sampled sound.

The Emulator II's keyboard algorithms provide a variety of new voice assignment techniques. Any number of different voices (limited only by available memory) may be assigned to the keyboard simultaneously. Each sound may have its own independent keyboard range, or may stack with or overlap other assigned voices. Up to eight different sounds may be assigned to a single key. Also featured: eight independent channel outputs, the ability to assign a different voice to each channel, and a track-oriented digital sequencer with auto-correct and extensive editing capabilities.

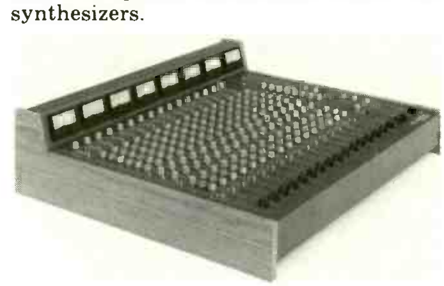
An RS-232 interface provides for enhanced computer access and control; MIDI allows connection to other synthesizers and sequencers, and, possibly most important, a built-in SMPTE reader/generator is provided.

E-MU SYSTEMS, INC.
2815 CHANTICLEER
SANTA CRUZ, CA 95062
(408) 476-4424

For additional information circle #136

SPECK UNVEILS SPECKMIX MK II CONSOLE

The new Speckmix Mk II 16-8-16 incorporates recording console three-band parametric equalization, two pan pots, cue and echo sends, solo, mute, and the ability to select between two line input signals a feature said to be very convenient for studios that operate with digital drums and synthesizers.



The output section consists of three stereo output feeds, dual echo returns, communication facilities, and +4 DBM output reference on the eight bus and stereo feeds. Offered as an option and additional cost are Jensen mike pre-amps.

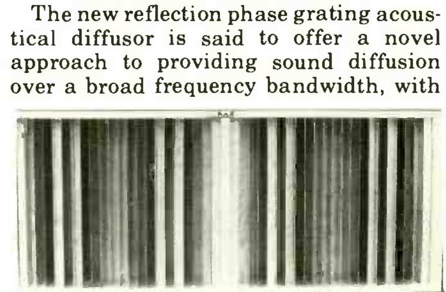
Suggested price of the Speckmix Mk II is $4,375.

SPECK ELECTRONICS
12455 BRANFORD STREET
ARLETA, CA 91331
(818) 897-4188

For additional information circle #137

RPG DIFFUSOR SYSTEMS UNVEILS NEW ACOUSTIC TREATMENT

The new reflection phase grating acoustical diffusor is said to offer a novel approach to providing sound diffusion over a broad frequency bandwidth, with



uniform wide angle coverage. The RPG™ can be used to furnish the necessary diffusion in the live end of a LEDE™ control room, and to help maintain a uniform stereo perspective across the entire width of the mixing console.

In addition, the material can be used to improve the acoustics of small isolation rooms, mobile studios, and drum booths, by greatly increasing the number of reflected wavelets, thereby adding ambience and body to the sound.

In the studio, it can be optimally mounted on the walls and ceiling to tailor the overall diffusion and reverberation or used as a movable gobo, allowing the acoustical characteristics of selected "live" areas, reserved for recording acoustic instruments, to be easily altered.

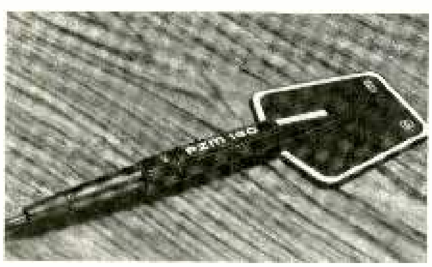
The RPG is custom designed with a computer-aided design and evaluation program, and the units are available in kit form.

RPG DIFFUSOR SYSTEMS, INC.
12003 WIMBLETON STREET
LARGO, MD 20772
(301) 249-5647

For additional information circle #138

CROWN ANNOUNCES PZM-180 PRESSURE ZONE MICROPHONE

The PXM-180 is a general-purpose microphone suitable for applications such as conferences, interviews, broadcast news and sports, and music recordings. Like other pressure-zone microphones, the PZM-180 utilizes a miniature condenser microphone capsule arranged very close to a sound-reflecting plate or boundary. This design is said to eliminate phase interference between direct and reflected waves for clearer, more natural reproduction.



An integral handle allows the microphone to be hand-held, stand-mounted, or simply laid on any hard surface. The PZM-180 can be phantom- or battery-powered; unlike some other PZMs, the new unit requires no external power-supply interface. Output is balanced, low impedance, available at an integral XLR connector.

Suggested retail price for the PZM-180 is $169.

CROWN INTERNATIONAL
1718 W. MISHAWAKA ROAD
ELKHART, IN 46517
(219) 294-5571

For additional information circle #139

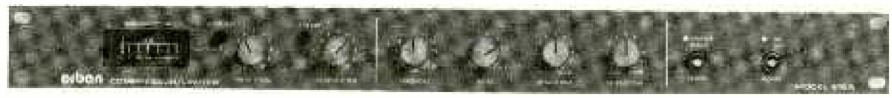
ORBAN MODEL 414A AND 412A COMPRESSOR-LIMITERS

The new Model 412A (mono) and 414A (dual-channel/stereo) are streamlined versions of the 422A/424A gated compressor, limiter and de-Esser, and feature user-adjustable Compression Ratio, Attack Time, Release Time, and Threshold controls, in addition to Input and Output attenuators.

Peak limiting and compressor functions are crosscoupled to eliminate potential pumping and modulation effects. The Threshold control with 20 dB range allows the user to determine the level at which gain reduction first occurs, without changing below-threshold gain and without compromising headroom or signal-to-noise ratio.

The new units use feedback control circuitry adapted from the Optimod-FM Model 8100A broadcast processor. The result is described as a straight-forward level control device that provides 25 dB gain reduction range with minimal audible side-effects, yet control range is adequate to produce special effects in production, if desired.

An illuminated, true peak-reading meter provides metering of total gain reduction, while two LEDs indicate program amplifier clipping and overload of the control circuitry.



Suggested retail price of the 412A is $425, and the 414A sells for $799.

ORBAN ASSOCIATES, INC.
645 BRYANT STREET
SAN FRANCISCO, CA 94107
(415) 957-1067

For additional information circle #140

New Products

VIDEO EDITOR INTERFACE FOR MCI JH-800 CONSOLE

The new Video Editor Interface has been designed to meet requirements for Video Editor Control of the JH-800 console. The unit routes crosspoint selection commands and ramp signals received from the editor to the JH-800. This is accomplished through an 8×16 matrix bus, through which the signal then controls the console channel or group that has been previously selected.

To maximize flexibility, internal matrix selector switches allow assignment of any crosspoint to any JH-800 channel or group. Ramp timing can be altered by two controls; one for turn-on, and one for turn-off.

The Video Editor Interface can offer audio system control from the Video Editor Edit Decision List, for facilities involved with audio for video production or posting. Factory preset matrix sets crosspoint one through eight to channel one through eight, respectively. Groups are not preassigned at the factory. Any console input fader, or group fader, may be assigned to the 8×2 matrix.

MCI/SONY PROFESSIONAL AUDIO PRODUCTS
PARKRIDGE, NJ 07656
(201) 930-6137

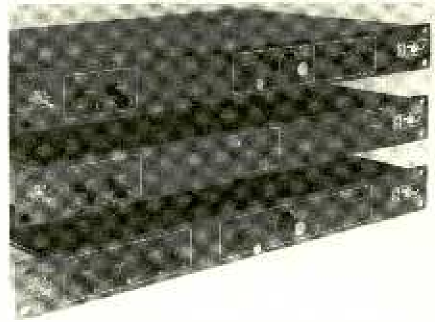
For additional information circle #143

AUDIO DIGITAL LAUNCHES TC SERIES OF DIGITAL PROCESSORS

The TC-3 pre-reverb digital processor is intended to operate with mechanical and digital reverb devices to enhance the sound by pre-delaying the audio signal. Because of its control versatility, the unit also will operate as a stand-alone digital delay. Delay times can be set to 1 second at a full 20 Hz to 20 kHz bandwidth, and 95 dB dynamic range. Suggested retail price is $599.

The TC-4, described as the lowest price broadcast profanity delay to enter the pro market, features 6.8 seconds delay time, with a full 15 kHz bandwidth and 85 dB dynamic range. The unit can be remotely bypassed to real time, or dead air at the user's discretion. Optionally, a user-installable "catch-up" card can be used to go from full delay, to memory dump, and back to a slow build-up of delay time. Retail price is $1,395.

The TC-5 industrial digital processor features variable delay times of up to 1 second at a full 20 kHz bandwidth.



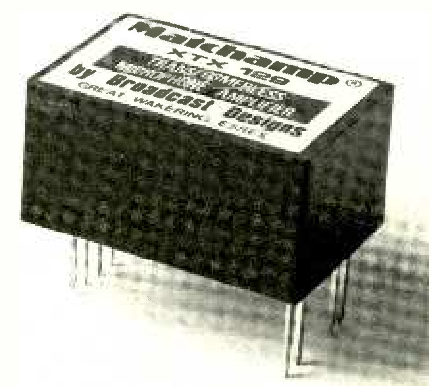
Intended to be used for echo cancellation, image enhancement and simple effect applications, the unit has a suggested list price of $649.

AUDIO DIGITAL, INC.
84692 SARVIS BERRY LANE
EUGENE, OR 97405
(503) 687-8412

For additional information circle #144

AUDIO + DESIGN/CALREC MARKETING XTX 129 MATCHAMP PRE-AMP MODULE

The XTX 129, now available in the US through Audio + Design/Calrec, Inc. is an ultra low noise amplifier housed in a small encapsulated block, with 10 termination pins on the bottom on a 0.1-inch standard spacing.



The specifications, which are said to approach theoretical limits, include relative input noise with 200-ohm source, 20 Hz to 20 kHz RMS of 129.3 dBv typical, and 129 dBv, guaranteed (IEC A-weighted of 131.3 dBv typical; and 131 dBv, guaranteed); CMR 100 dB typical (92 dB guaranteed) at 10 kHz, 110 dB typical (102 dB guaranteed) at 1 kHz, and 87 dB typical (82 dB guaranteed) at 60 Hz; plus distortion at 0 dB output better than 0.006%.

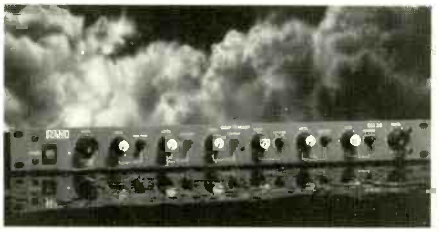
AUDIO + DESIGN/CALREC, INC.
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009/5010

For additional information circle #145

MODEL SM26 SPLITTER MIXER FROM RANE

The Model SM26 contains two master stereo inputs, six separate mono inputs, six separate mono outputs, and two master stereo outputs, all with separate level controls and mix/pan controls. Features include automatic balanced/unbalanced operation, minimum 12 dB overall gain, high-current output line drivers, and headphone-drive capability.

The unit can be operated as a 6-in/2-out line mixer with pan controls, a 2-in/6-out line splitter with mix controls, an 8-in/8-out line matching/booster amplifier for -10 dBV/+4 dBm interface, or a combination of the above.



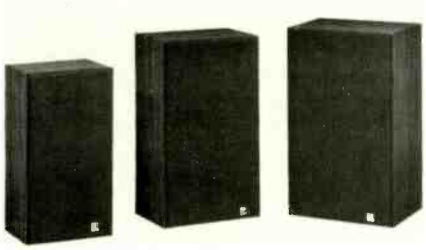
The Model SM26 carries a suggested list price of $299.

RANE CORPORATION
6510 216TH SW
MOUNTLAKE TERRACE, WA 98043
(206) 774-7309

For additional information circle #146

ELECTRO-VOICE LAUNCHES FIVE NEW COMMERCIAL SPEAKER SYSTEMS

The 100-watt PI 15-3 and PI 12-2 are described as highly efficient wide-angle systems designed for permanent installation indoors. Each system employs an EVM Series II low-frequency transducer, ST 350B controlled directivity tweeter and EV's HF auto-limiting circuit. Additionally, the PI 15-3 is fitted with the VMR vented midrange speaker. Vented bass cabinets are Thiele-designed and finished in black vinyl.



Also designed for indoor applications, the 50-watt Interface: 1, 2, and 3 are compact, wide-range speaker systems. The Thiele-designed vented cabinet has an 8-inch midrange/woofer and Super-Dome™ tweeter.

Suggested prices for the PI 12-2 is $405, and $655 for the PI 15-3. The Interface 1, 2, and 3 cost $199.94, $259.94 and $299.94, respectively.

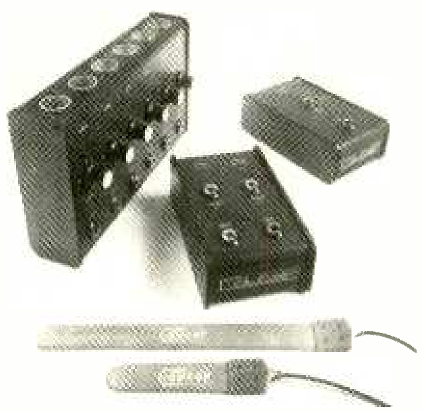
ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #147

New Products

C-DUCER MICROPHONE SYSTEM FROM C-TAPE DEVELOPMENTS

The new CX Series flexible microphone system is housed in a custom-designed extruded metal enclosure, and is supplied in a durable carrying case. The pre-amp features 79 dBA signal-to-noise level, and "exceptional" dynamic range. The units can be phantom powered, a C-ducer power supply being available for use without phantom power.



Both 600-ohm, transformerless balanced and high-impedance, unbalanced outputs are provided; output level is externally adjustable from mike to line level. One-, two- and six-channel formats are available, with the latter forward having application primarily on drum kits.

Listening tests on the new series are said to indicate several areas of improvement over the existing C-Series, which it replaces. The new two-Channel CX Series system carries a suggested resale price of $326, including two C-ducer mikes.

C-TAPE DEVELOPMENTS, INC.
P.O. BOX 1069
PALATINE, IL 60078
(312) 359-9240

For additional information circle #151

RENKUS-HEINZ UNVEILS THREE NEW LIVE-SOUND UNITS

The SWG20-3 three way, active crossover/signal processor offers 18 dB per octave continuously variable crossover, thermal and displacement overload protection, equalization, and time compensation. Crossover ranges are 250 Hz to 2.5 kHz, and 500 Hz to 5 kHz. The midband section optimizes use of the SSD5600 high output mid-range driver.

The SSA242 adaptor allows coupling the 2.4-inch throat of an SSD5600 to standard 2-inch horns. Capable of 100 acoustic watts output, the midrange driver can be retrofitted into existing systems to increase SPL, or used with a variety of R-H two-inch exit horns.

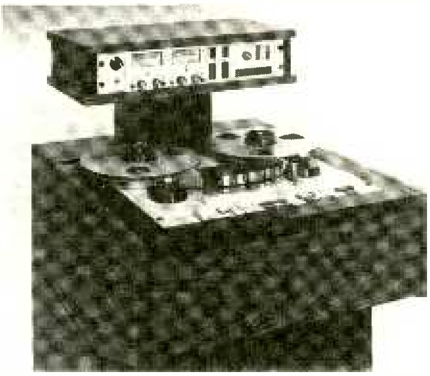
The CBH500 constant beamwidth horn features a two-inch throat that offers 60- by 40-degree constant beamwidth coverage above 500 Hz. Well defined horizontal and vertical patterns are said to make the horn ideal for mid- and long-throw applications, facilitating system and cluster design. An adaptor is available for one-inch throats.

RENKUS-HEINZ, INC.
17851 AB SKY PARK CIRCLE
IRVINE, CA 92714
(714) 250-0166

For additional information circle #152

L.J. SCULLY MODEL LJ-12 STEREO TAPE MACHINE

The new recorder features three advanced microprocessors that digitally control the transport and analog audio signals, and non-volitale memory of calibration, EQ, and bias settings.



Other features include glass-bonded ferrite heads, gold-plated connectors, lifetime hall-effect transport switching, SMPTE compatibility, and four-speed operation variable from 3 to 36 IPS in 0.01-IPS increments.

L.J. SCULLY MANUFACTURING
138 HURD AVENUE
BRIDGEPORT, CT 06604
(203) 368-2332

For additional information circle #153

ROCKTRON "HUSH" NOISE-REDUCTION SYSTEM

Hush is described as the first single-channel encode/decode noise reduction unit, designed for instrument and line level inputs, to eliminate noise in countless signal processing applications, including tape echo and delay systems, plus reverb chambers.

Operating over the entire audio frequency range, the unit provides an effective noise reduction of greater than 30 dB. Continuing in the tradition of previous Rocktron noise reduction designs, the Hush is said to process audio signals without the addition of objectionable side effects like distortion, noise, or "pumping."

The device employs a two-step process, compressing the dynamic range upon encode and expanding it upon decode, providing dynamic range in excess of 110 dB.

ROCKTRON CORP.
2146 AVON INDUSTRIAL DRIVE
AUBURN HEIGHTS, MI 48057
(313) 853-3055

For additional information circle #154

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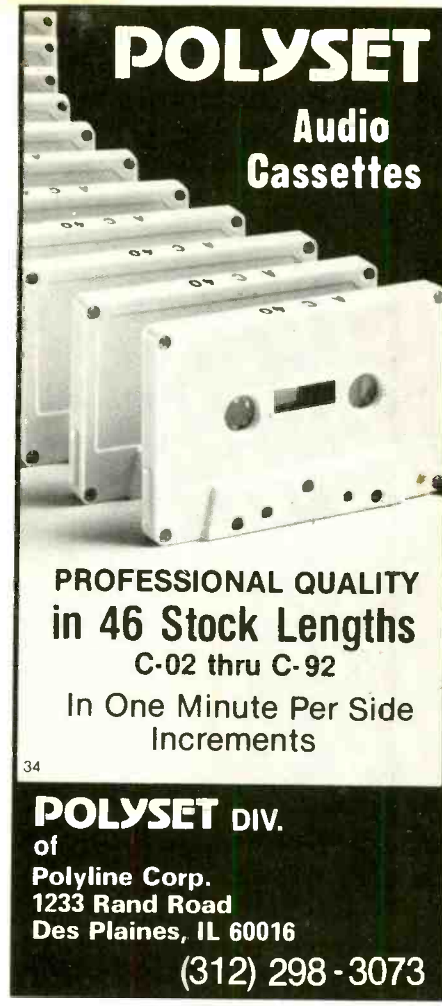

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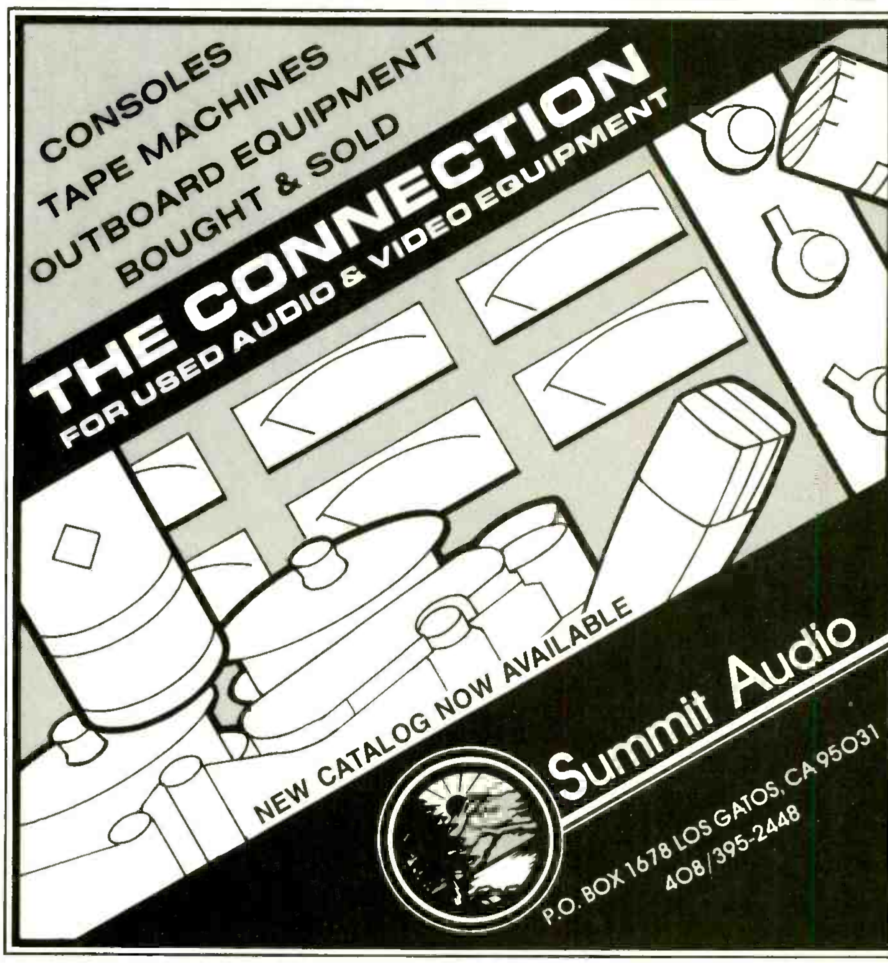

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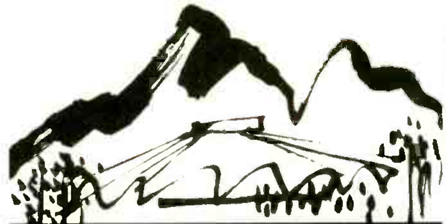

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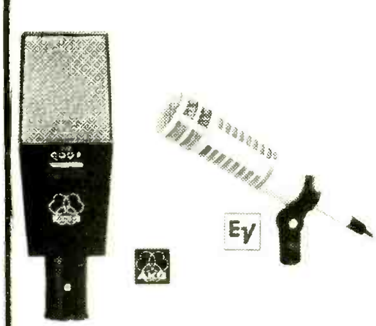

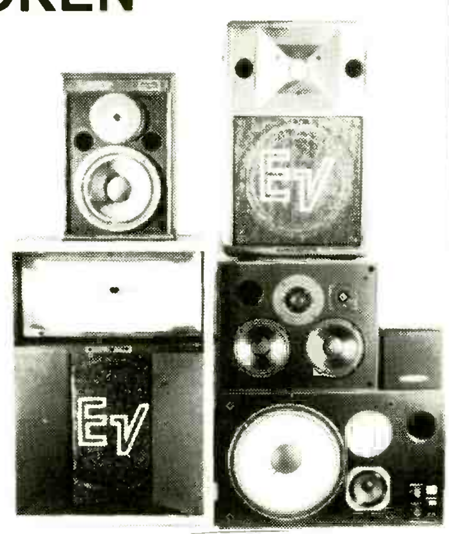

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For additional information circle #162

news

— continued from page 55 . . .

700, which utilizes dbx's proprietary CPDM (Companded Predictive Delta Modulation) technology, to be used by radio stations for broadcast applications, as well as by recording studios. "Several broadcasters, including RKO Radio Networks and WGBH Radio [Boston], are already utilizing the processor for high-quality point-to-point transmission," Korthals added.

On March 23 WGBH began transmitting, via digital audio, live broadcasts of the Boston Symphony Orchestra. According to Anita McFadden, operations manager of the PBS station, WGBH will be using two dbx Model 700 Digital Audio Processors to send weekly Boston Symphony concerts direct from Symphony Hall to its transmitter site in nearby Milton. According to McFadden, "This experiment has two unique aspects: 1) the dbx 700 will be the first digital audio processor used to improve a studio-to-transmitter link; 2) This will be the first use of the dbx 700 for a live remote pickup of a concert performance."

The station conducted unannounced tests of the system in December and January. "Not only was the absence of noise remarkable," remarked chief operator Peter Swanson, "but the absolute flatness of frequency response of the dbx system prompted one of WHBH's severest critics to call up the station and [comment] 'Best symphony broadcast I've heard in 20 years.' During the 'secret' test of the system, other listeners called spontaneously to comment on the unusually clean and live-sounding broadcast."

A team headed by David St. Onge, chief engineer of WGBH TV/FM, set up microwave links between Symphony Hall and Boston's Prudential Center, where sister TV station, WGBH Channel 2, has a video repeater to the station's studios, since there is no line-of-sight path for a single "hop," according to McFadden. Another microwave path was established between the WGBH studios and its Milton-based FM transmitter site.

"Usually, our studio-to-transmitter-link path provides a signal-to-noise ratio of only 65 dB with Dolby-A noise reduction," McFadden noted. "In our end-to-end test with the dbx system, we measured signal-to-noise ratio in excess of 80 dB, plus another 20 dB headroom."

For the unannounced broadcasts subsequent to the tests, a dbx Model 700 processor was set up for encoding from Symphony Hall, while a second Model 700, at the transmitter site, was used for decoding.

DEC OPENS NASHVILLE OFFICE

Digital Entertainment Corporation has opened its latest sales and service facility in Nashville, to be headed by Mid—America regional sales manager Thomas Behrens, formerly an engineer with The Bennett House Recording Studio and Valley People.

"We were keenly aware that the Mitsubishi Digital Audio System had many friends in Nashville, and felt that we could best se. ve our clients all over the Central

news

United States by creating a new sales and service facility there," explained DEC president Tore Nordahl. "The Nashville studios are ready for digital audio equipment and we wanted to express our serious commitment to them in a very real way."

The new office is located at: 2200 Hillsboro Road, Nashville, TN 27212. (615) 298-6613.

CALREC AUDIO PARTNERS AUDIO+DESIGN

Just before Christmas, Audio+Design Recording, Ltd. and Calrec Audio, Ltd. concluded an agreement whereby Calrec acquired a substantial share-holding in the USA subsidiary Audio+Design Recording, Inc. based near Seattle, Washington. As part of the same agreement, Nigel Branwell, former VP of the US corporation, becomes president of the new company now known as Audio+Design/Calrec, Inc.

According to Branwell, "This move heralds a period of significant growth for the US company, and could not come at a better time, coinciding as it does with real signs of recovery in the USA. Calrec has established an excellent reputation, and there are more high technology products on the way. Initially we shall be concentrating on the microphone range, including the Mark IV Soundfield Microphone, and top-of-the-range portable broadcast mixers. As soon as we have established an effective way of offering Calrec's custom services to clients here in the USA, these will be made available together with the company's new Assignable Mixing Consoles for both recording and broadcast use."

UNIVERSITY OF IOWA SUMMER AUDIO SEMINAR

The University's Seminar in Audio Recording runs from June 18 thru 29, and is offered to qualified people for instruction in professional techniques of sound recording.

The seminar provides intensive training in four-hour sessions, under the instruction of Stephen F. Temmer, and includes daily classroom/studio lectures, plus hands-on familiarity with Neve and Langevin mixing consoles; Studer, Ampex, Crown, and ReVox recorders (two-, four-, and 24-track); AKG, Neumann, and Shure microphones; EMT 140TX plate and EMT 251 digital reverberation systems; UREI equalizers; Dolby dbx, and Telcom noise reduction systems; McIntosh, Crown, and JBL monitoring equipment.

Discussion topics will include the fundamentals of acoustics; a special emphasis on microphone theory; VU and peak program meters; and emerging technologies, encompassing the fundamentals of digital recording and direct metal disk mastering. Also, comparative demonstrations of equipment and techniques will be emphasized throughout the program.

For further details, contact Lowell Cross, School of Music, The University of Iowa, Iowa City, IA 52242. (319) 353-5776/353-5977.

DELTALAB FILES PATENT INFRINGEMENT SUIT

DeltaLab Research, Inc. recently filed suit in San Francisco Federal Court, 1984, against ADA Signal Processors of Berkeley, CA for patent infringement. According to company president Richard E. DeFreitas the action was taken as a f
step toward protecting the tec
DeltaLab has developed over
eral years. Simply stated, he
icy is as follows: "We took de
tion out of the dark ages, a
suitable for high-quality audio
welcome fair competition from c
nologies such as PCM, we canno
unfair use of our own technology
who would exploit our patent righ

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will cover reverberation, sim
ambience, cognitive processes in
composition, real-time systems, p
acoustics, and D-to-A conversion.
lecturers include Barry Blesser of th
Max Mathews of Bell Laboratori
nard Gordon, president of the Ar
Corporation; and John R. Pierce o
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For application information, c
Director of the Summer Session,
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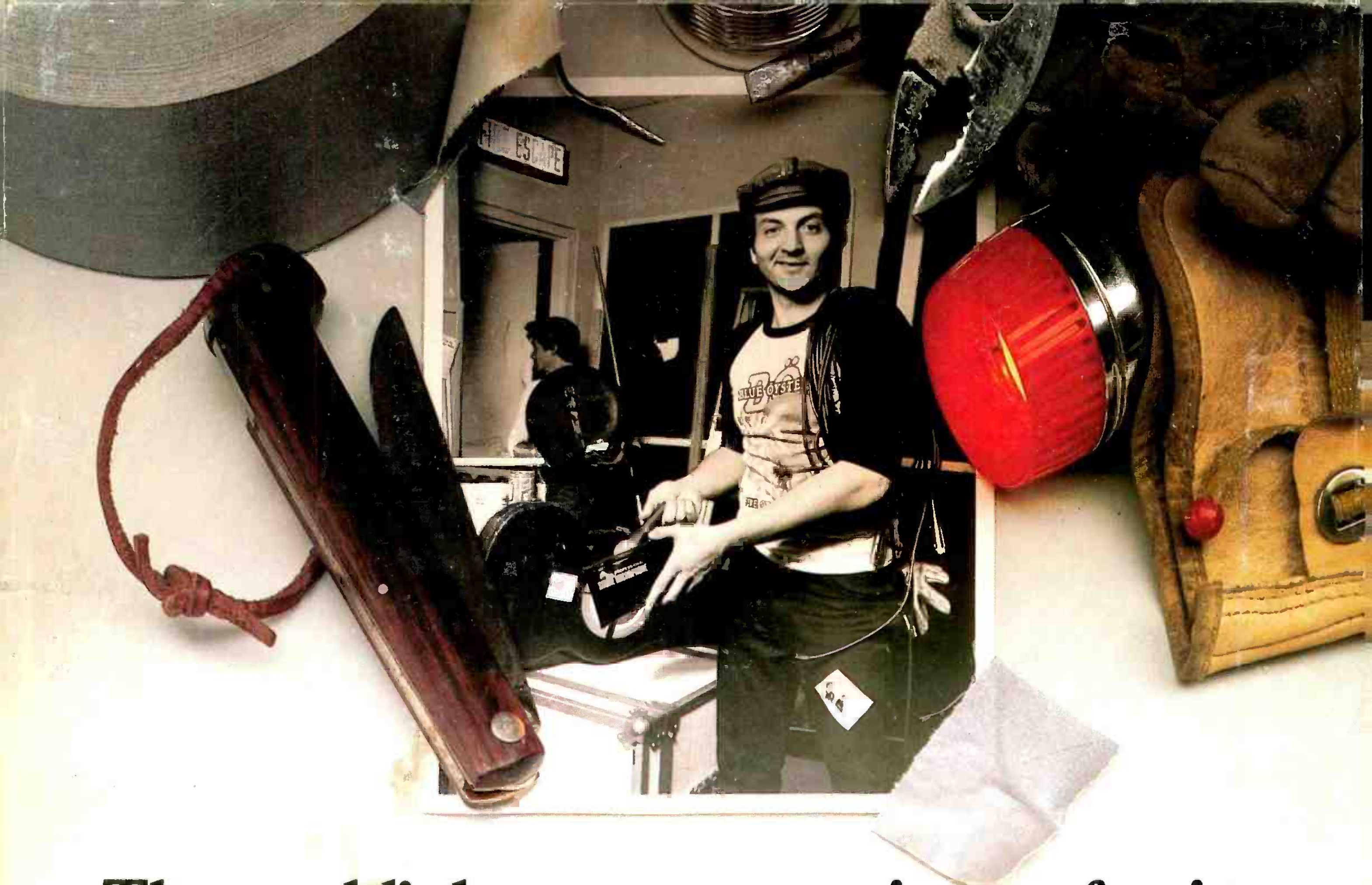

The world's least conservative profession has maintained one rigid tradition. The SM58.

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.

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Simply because there is no sound quite like the SM58 sound. Its punch in live vocal situations, coupled with a distinctive upper midrange presence peak and fixed low-frequency rolloff, give it the trademark quality no other manufacturer can imitate, although others have tried.

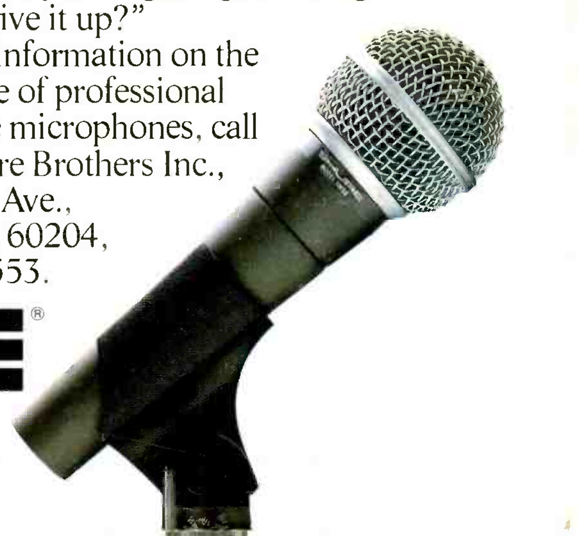
And to protect that sonic perfection, the SM58 is extraordinarily tough. Even six-foot drops on hardwood floors won't faze it. Ask any roadie who has used–and abused–one.

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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