RECORDING ENGINEER/PRODUCER

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

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As designed for Motionpicture Recording, Inc. in Hollywood, the AMEK M-4000 CINEMA console offers simultaneous dubbing to mono, stereo 3 track, 3 into 5, 4 track, 6 track Dolby stereo, and 8 track Omnivision, with additional 24 track output assigns and mic inputs for Foley recording.

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For additional information circle #1

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For additional information circle #3

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

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

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August 1983
Volume 14 — Number 4

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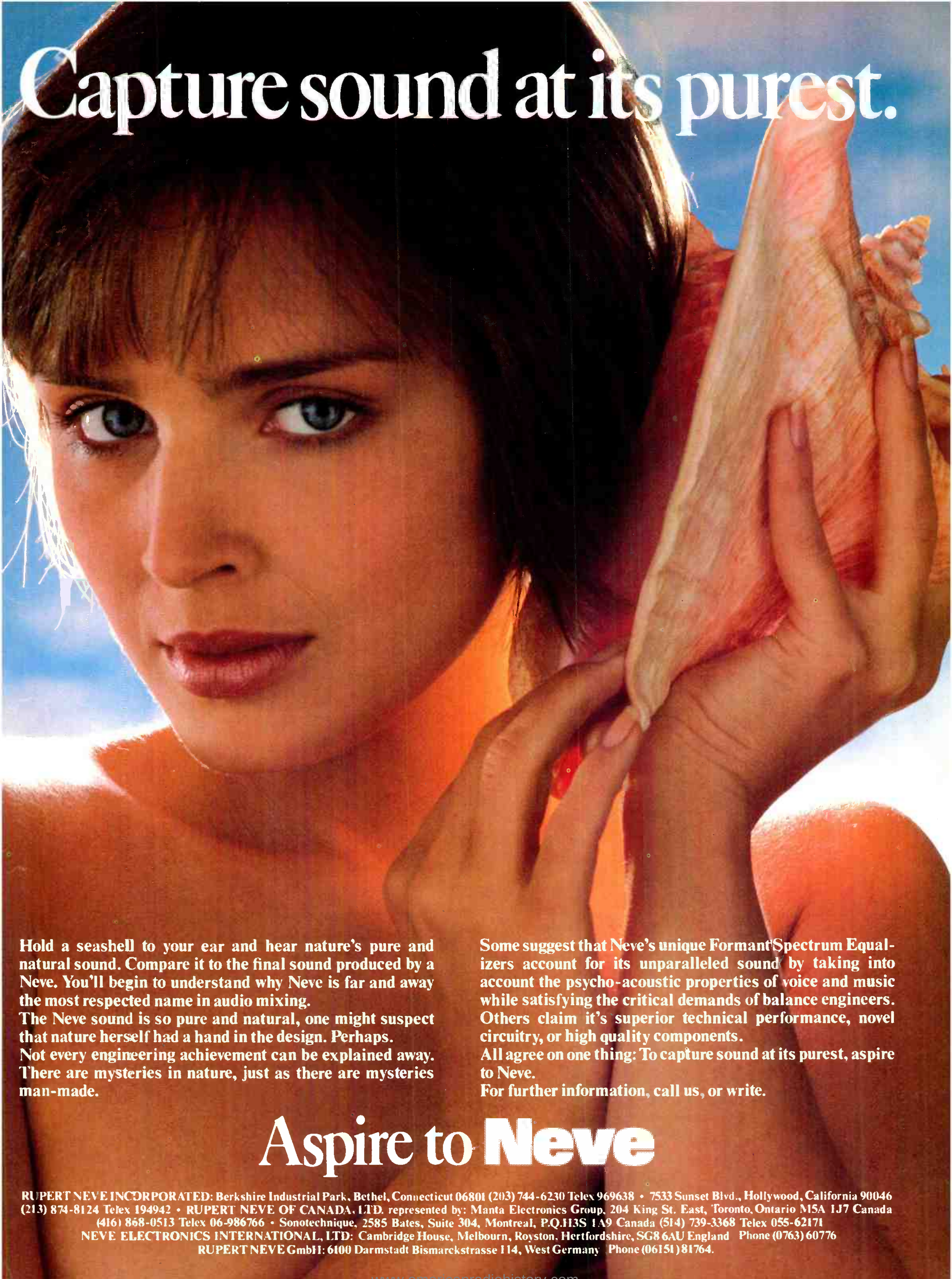
Interior of **Westlake's new Studio C** control room, featuring an electrically operated skylight, and innovative Controlled Travel Path™ acoustics design, described in a feature article on page 50. Photography by **Kathy Cotter.**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The features don't stop here; 8X Series consoles also include super solo sections (giving instant access to pre-fader, post-fader and tape solo), comprehensive slate and talkback systems, a built-in calibration oscillator, and a high speed LED metering array in an easy-to-read meter bridge assembly. Standard module features include XLR balanced inputs (both mic and line), XLR balanced outputs (buss and stereo master outs), continuously variable mic and line input gain controls, switchable phantom power, phase reverse, pad, 12dB/octave high pass filter, EQ bypass switch, channel on button (w/LED indicator), channel peak clip LED, and the exclusive Audioarts Engineering M-104 precision conductive plastic linear fader.

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For additional information circle #5

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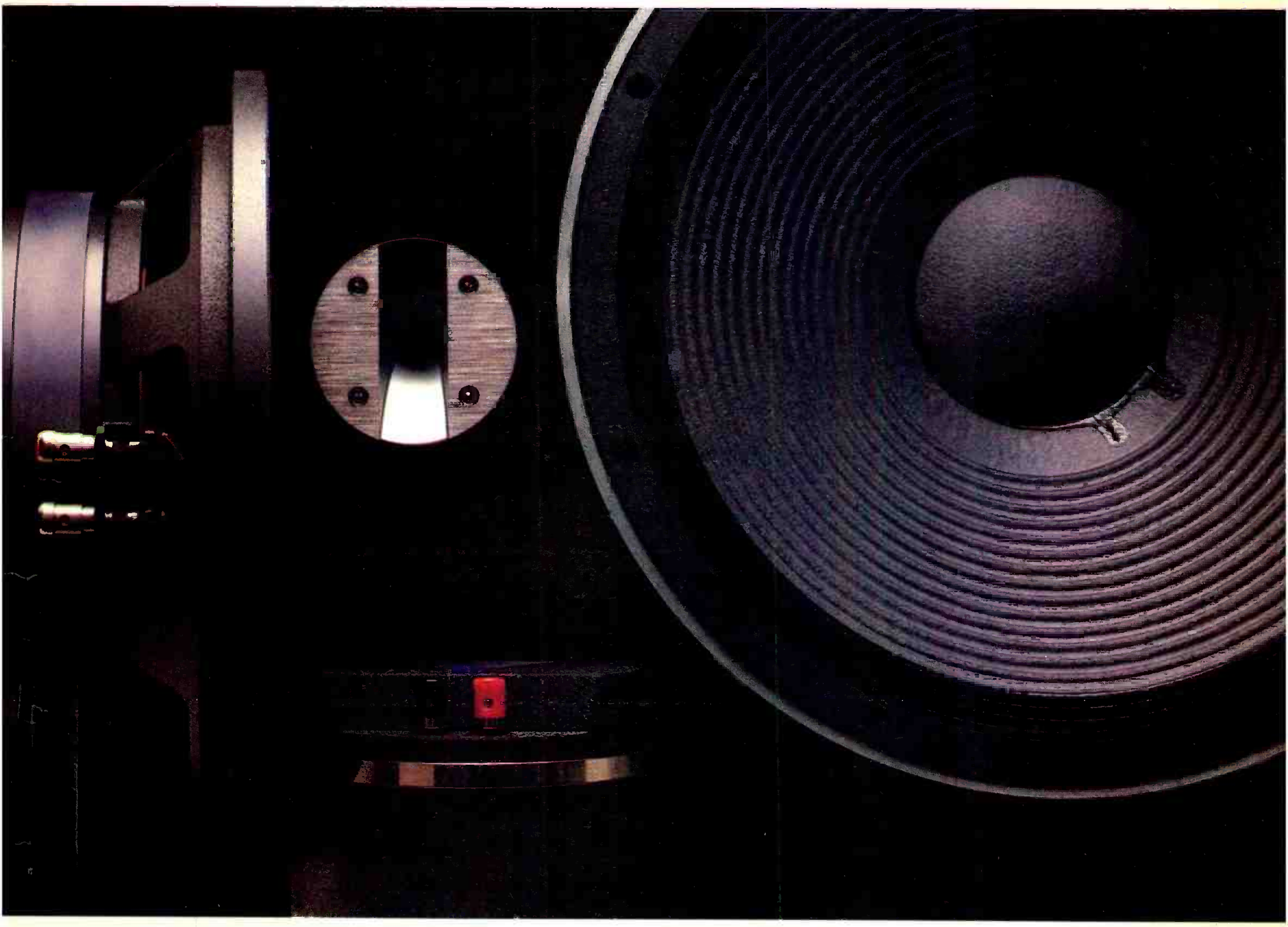

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For additional information circle #6

views letters news

AUDIO ARCHIVE UPDATE

from: **Bill Storm, director**
Belfer Audio Laboratory
and Archive
Syracuse, New York

The article entitled "A Perspective on Recorded History — The Audio Archive and Laboratory at the University of Syracuse," by David Moore, published in the April 1983 issue of *R-e/p*, was appreciated, and has stirred a lot of interest.

A reader and colleague has asked that we clarify two points:

1. *Aluminum disk reproduction.* Properly fitted diamond styli can reproduce sound from aluminum disks; however, at risk to both stylus and disk. Unless you're prepared to invest in the proper styli, and understand the inherent problems, we would not generally recommend diamond. Fibrous or thorn styli would be safest.

2. *Cylinder record composition.* The earliest record to use a tube coated with wax was the Bell-Tainter graphophone cylinder. This was basically an experimental medium demonstrated primarily for dictating purposes. The first general-usage cylinders were homogeneous wax, and not "coated" tubes.

A cardboard insert was later used by Indestructable Records of Albany, New York. These records used the cardboard insert with metal rings at each end inside, not a wax, but a celluloid compound originally patented by Lambert.

A celluloid material was also used in the famous Edison Blue (and later Purple) Ambersol cylinders, but without cardboard inserts.

Another point we would like to clarify is that the variable acoustics system in the [laboratory's] studio is used primarily to vary reverberation and absorption characteristics for experimental purposes. Simulating different listening environments is but one small facet of the system's use.

Again, thank you for a job well done. In two such subjective fields as music and sound recording, we hope we can try to stay objective in the way we save and protect this material for posterity.

ON THE RIGHT TRACK

from: **Charlie Morgan, president**
Morgan Sound, Inc.
Lynnwood, California

I thought I'd take a few minutes to write and tell you that I really appreciate the fine articles you are running on live concert sound in *R-e/p*. As you are well aware, the recording industry is setting new goals and presenting new challenges for those of us doing live audio on the road. A good recording dictates the necessity for comparable quality audio in live performances. The need for idea sharing, and an open approach to techniques, has never been more important than now in helping to stimulate the live performance aspect of our business.

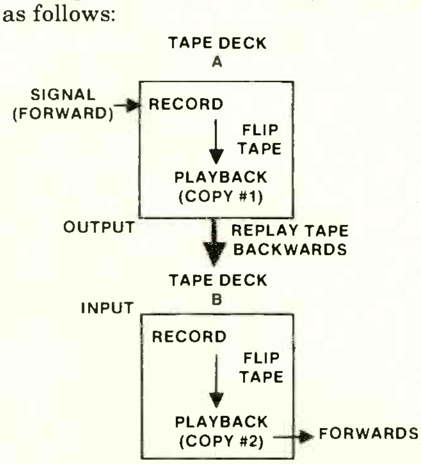
I just finished reading the latest article on the new Audio Analysts system out with Styx [written by David Scheirman], and have to say: "Well done, Dave!"

AUDIO MYTHOLOGY: BACKWARDS TAPE DUBBING

from: **Douglas Pomeroy**
Pomeroy Audio
Brooklyn, NY

In John Robert's column, "Exposing Audio Mythology," published in the April 1983 issue of *R-e/p*, he lists as one of many possible topics for examination the assertion that dubbing tapes backwards improves transient response of the copy. Perhaps he has seen J.W. Beauchamp's letter in the *Journal of the AES* (1968, page 112), which states that "the Fourier transform of a time-reversed signal is the complex conjugate of the signal in forward time." The requirement is that the tape recorders used be as nearly identical as possible with regard to frequency response, etc.

The procedure, as I understand it, is as follows:



As can be seen, phase correction can take place only at the expense of an additional generation of tape copying; the other degradations of quality associated with dubbing would, of course, continue to multiply.

To complete all the steps necessary for maximum phase compensation, final (forward) playback must be via one of the identical tape recorders, and this, if true, seems to me to greatly reduce the practical usefulness of the whole procedure.

Never having put the scheme to a test, I'd be very interested to hear from any engineers who have seriously tried it. (I believe that Tony Faulkner, working in England, is one such person.)

Meanwhile, I found excellent your discussion of the matter of absolute polarity.

John Roberts replies:

Thank you for your kind comments. On the subject of "backwards dubbing," I must admit I am not very enthusiastic. I suppose if you happen to have a few tape recorders lying about that don't mind running backwards, and you need dubs for playing back on the same model machines, then by all means go for it. (Note: don't use your noise reduction — the attack and release times will be scrambled.) Backwards dubbing in this controlled circumstance does appear to offer a first-order phase correction. However I wouldn't suggest rewiring you high-speed cassette duplicators to run backwards just yet.

It might be interesting to see what this technique could do for the rather severe low-pass filtering found in consumer digital players. While there is no standardization of filtering between players, I expect some *de facto* standards could arise should this technique actually do something audible. If someone out there has some first-hand experience with this, or has any observations to make, please drop *R-e/p* a line with your comments. I wonder if I could get the Hartford symphony to play Tchaikovsky backwards?

ABSOLUTE POLARITY

from: **Norm Laramee**
Director of Engineering
KLAZ-FM/KOKY-AM
Little Rock, Arkansas

Regarding John Robert's article on absolute polarity [published in the April issue]: I have yet to encounter any noticeable negative effects resulting from not observing absolute polarity — with one exception, however, that being the effect mentioned by Don Hobson of KJQY-FM in your June issue ("Letter to the Editor"]. In the broadcast environment, it is *essential* that absolute polarity be observed if the anouncers are going to monitor their broadcast via a receiver. The reason is quite simple [since] the announcer is hearing his (or

— *continued on page 16 . . .*

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Actual Music Printing Sample, Reduced

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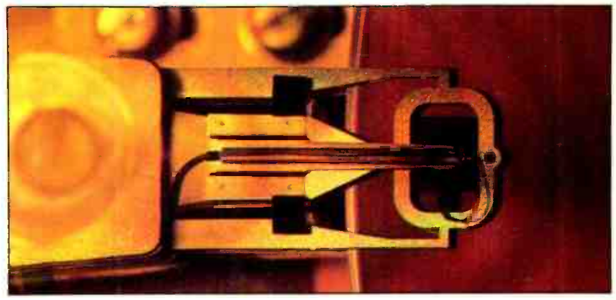

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LETTERS

. . . continued from page 12 —

her) voice from three sources:

1. Through the bone structure of the head.

2. Acoustic leakage of room ambience around the headphone cups.

3. Electrically through the headphone transducers via the receiver.

The first two sources are always in phase; it's the third source that can be a problem. Naturally, if the received signal is out of phase with the first two, there will be some degree of cancellation perceived only by the person speaking. I've had it happen to me simply by changing microphone or a mike cable or, more commonly, [while] experimenting with equipment in the audio chain. I have found that some audio processing devices will give an output signal that is inverted from the input. Looking at the schematics, it's easy to pinpoint the [signal] inversion. In fact, on some devices that I have observed the polarity is flipped two or three times before being outputted.

The reasons for this are obviously due to a necessity in circuit design, and do not take away from the equipment's performance. This effect, by the way, is not exclusive to FM. The potential for absolute polarity reversal is just as great on AM stations; the effect is identical. I remember when it was popular to use a device called "Symetripeak" on AM stations. Its function was to constantly switch polarity of the audio signal to the transmitter, to transform naturally asymmetrical voice to a symmetrical form. It caused all kinds of headaches (literally) for announcers. Most engineers at the time, I recall, could not figure out why!

There are some other devices still in use today that feature some form of dynamic phase reversal: the BL-40 Modulimeter, for instance. Since I do a fair amount of on-air testing of new types of audio processing equipment, about four years ago I installed a polarity reversal switch in the microphone line in the Control Room. So, whenever I install a new piece of audio gear, and the announcer has difficulty hearing, they just flip the switch and are back in business again.

GRATEFUL DEAD SOUND SYSTEM DESIGN

from: **Dave Trautman**
CTEN
Edmonton, Alberta
Canada

There's a story I love to tell my friends about a trip I made to Vancouver when I was younger. An acquaintance had laid a free ticket on me for the Grateful Dead, with the only hitch being that I had to get to Vancouver to hear them. Spending my last hard-earned dollar, I managed to appear at a packed PNE Colliseum to be at my first Dead concert. I have been a devoted fan ever since.

I bore my friends with my enthusiastic descriptions of the sound system that I heard, and the unbelievable clarity of that performance. I have tried to describe to them what I thought I saw on stage, but did not understand — it was a "wall" of speakers 40 feet high. It wasn't loud, but incredibly *clear*; I swore that I could hear whispering on stage during quiet passages.

For 10 years I have wondered how it was that the Grateful Dead could have such a magnificent sound, and nobody else could. After reading David Gans' story about 20 years of experimenting with the Grateful Dead [June 1983 issue, page 98 — *Ed.*], I finally was able to find out just what was so special about that PA system.

I want first to thank *Recording Engineer/Producer* for publishing the story, and artist Mary Ann Mayer for the sketch, so that one of my life's little mysteries could be cleared up.

Upon finishing the article, I couldn't help wondering why more groups don't use the same system. It appears much more simple to operate; it is artist friendly; logical; free from monitors;

and, above all else, truly transparent.

David Gans explains that the Dead's mixer, Dan Healy, found the system too big, too cumbersome, and too labor intensive to continue with, and adds that newer designs have improved on conventional PA systems. But after watching Supertramp assemble a huge stage, lighting, monitor, projection, and PA system, I can't help wondering if the "Dead" PA wouldn't make them happier. I think that it was the fast pace at which the group was trying to perform that killed the idea.

I hope that some of the better ideas from that "wall of speaker" system, with its microphone-pair vocal concept, will be picked up by others and used, because I for one am very tired of not hearing the band at a concert.

WHO MULTIMEDIA

from: **W. Mark McKibben**
VP Engineering
Wold Communications, Inc.
Los Angeles, California

I was very interested to read Paul Lehrman's article in the June 1983 issue of *R-e/p*, concerning the Who concert broadcast on December 17 of last year. As he stated in his author's note, this was an extremely complex network.

In reading the article, I did note one minor error on page 94. At the bottom of column one, the author refers to "AT&T's Westar IV." Naturally, I'm sure he meant to say *Western Union's* Westar IV.

I would also like to point out that Wold Communications was the prime contractor for the communications ordered in connection with this event. Although the author mentioned Wold many times in the article, it does not convey the extensive role played by our company in designing and accomplishing the network. In putting this together, the following entities were simply subcontractors to Wold: Canadian Teleconferencing Network; Telsat Canada; Canadian Bell; Satellite Signals Unlimited; AT&T; and several other entities.

On page 95, column one, paragraph two, the fact that Wold was a central focus point for this event was alluded to in the article.

news

NEW AUDIO SERVICE FACILITY FORMED

Hy James, Inc. and DLC Design, of Farmington Hills, Michigan, have jointly formed a professional audio service facility, to be known as Electro-Media Service. Located within the Hy James complex, EMS will offer ben and field recording equipment repair, and complete studio maintenance programs. The new facility will be headed by former U.S. Pioneer service manager, Jim Pashkot.

For further information contact: Electro Media Service, 24166 Haggerty Road, Farmington Hills, MI 48018. (313) 477-6502.

DATATRONIX TO BEGIN DIRECT MARKET STRATEGY

The company will now market its audio and broadcast product lines directly from its manufacturing facilities in the greater Washington, D.C., metropolitan area, rather than through the distributors Datatronix has used in the past. According to Datatronix general manager George J. Ripol, "We believe we can afford faster and more comprehensive services to our customers through direct contact with our engineers and audio/broadcast specialists who are sensitive to the needs of the industry."

The company, which manufactures API mixing console, TELEPATH intercom systems, and a complete line of modular components, is retaining the services of Studio Consultants, New York, and Nissho Iwai, Tokyo, and is actively seeking additional foreign representation.

— *News continues on page 129 . . .*

Studio Applications of the RTW Analog-to-Digital Interface Unit for Sony's PCM-F1 Digital Audio Processor

by Larry Lamoray
International Marketing Manager, Auditronics, Inc

There appears to be a certain degree of confusion in the pro-audio marketplace regarding the intended use and function of the recently introduced RTW Studio Processor Set, currently being marketed exclusively in the U.S. by Auditronics, Inc. While, to date, the interest shown in the unit far exceeds the manufacturer's expectations, some misunderstandings exist as to what the RTW Processor does, and does not do. [For a description of the unit's main features, see the "New Products" item on page 115 of the June issue — Ed.]

While the Sony PCM-F1 Digital Audio Processor is an excellent piece of equipment (and an exceptional value), it is unfortunate that the unit originally was designed for, and targeted at, the consumer market. As a result, the digital code format of the F1 is similar to, but non-compatible with, that of Sony's larger professional digital audio systems, including the PCM-1610. So, although interest in using the F1 as an inexpensive vehicle for high-quality audio grows, the facts that there are no editors currently available for the F1 format, and that Compact Disc mastering requires that U-Matic master tapes be prepared to the professional 16-bit format with accompanying timecode tracks, dictate that use of the PCM-F1 be limited, or at best compromised, by a lack of available compatible hardware.

The RTW A/D Interface was developed originally at the request of a few European broadcasters that possessed Sony PCM-1610 Digital Audio Processors with accompanying editing systems, and who wished to utilize the F1 for Electronic News Gathering or Field Production, and then be able to edit and assemble in-house on their larger system. It so happens that this same philosophy of use is applicable to recording studios wishing to enter the digital audio age, and master for Compact Disc. However, the RTW unit is *not* a panacea for all of the problems involved.

The RTW unit will translate PCM-F1 14- or 16-bit data to PCM-1610 format within the digital realm, for direct digital copying and subsequent editing; *no* conversion to analog format is required. However, a PCM-1610 Processor must still ultimately be involved to edit, or to master for Compact Disc.

The unit will provide balanced, line-level analog inputs and outputs via XLR connectors, and 10 dB of headroom optimization by inverse gain control in 2 dB steps. The PCM-F1 alone has unbalanced inputs and outputs, at -10 dB, via phono-type connectors. The RTW Interface also will provide DC powering for the PCM-F1 and up to two VCRs, such as the Sony SL-2000 Betamax, and will interface between them for direct digital-to-digital copying. In addition, it will provide a more comprehensive display of the error correction taking place within the PCM-F1, together with manual control of the F1's pre-emphasis function.

The RTW unit is similar in physical size and appearance to the Sony PCM-F1, and is available with transformer-isolated or active-balanced differential inputs and outputs. An accompanying modification is required to the F1 being used with the unit, and it should be noted that this modification will void the warranty. ■■■

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When you know exactly what you want but your console keeps saying, "You can't get there from here", there's something you can do besides screaming and pounding your fists. Haul yourself down to your TASCAM dealer and take a hard look at our M-16.

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For additional information circle #12

EXPOSING AUDIO MYTHOLOGY

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

by John Roberts

In response to my column in the April issue dealing with "Golden Ear" terminology, I have received the following "Not-quite-so-GE-Glossary":

1. *Imaging*: Hallucinations caused by too much Tequila.

2. *Soundstage* (soundspace): A straight line between the speakers.

3. *Brittle metallic sound:* One of the tweeters is rattling; the back panel of the console is loose, and about to fall on your toes.

4. *Coloration:* Pure, regular distortion.

5. *Hard, Edgy:* Regular distortion made nastier by higher level; the producer is deaf at 3.6 kHz.

6. *Tonal purity:* You've left the desk line-up oscillator on; you are playing the master tape before the first track where the tones are.

7. *Hard, hardening and coarsening of texture:* The producer is going deaf at 3.6 kHz; you hit your head on the ceiling slung monitors; the vocalist is pulling rocks out of the acoustic treatment, and throwing them at you.

8. *Thicken texture:* Put back the front covers on the monitors.

9. *Definition:* Ability to identify accurately which of 16 drum mikes you're listening to.

10. *Weight:* One of the monitor has worked loose, and fallen through the floor into Studio Two.

11. *Congestion:* You have left up in the mix at least one of the badly-recorded guitar solos on tracks 14, 17, 21, 22, 23, and half of 24 (with the second part on track 2), along with the real one (which is on tracks 13, 16, 20, and the first half of 24) by mistake, and it (they) is (are) getting in the way.

12. *Cool sound*: You are working on a Jazz album; you just walked in from the Jacuzzi; it is winter here in Boston and you left the studio door open — snow is piling up on the console, and the heating system is about to burn out.

13. *Harmonic richness:* Nice distortion (see #4).

14. *Dynamic contours:* The producer is messing with the master fader; the compressors are not working; the VU meters move more than 0.1 dB during the track.

15. *Tube-like roundness:* Fashionably nice distortion (see #13).

16. *Character:* A room that wasn't designed by an "acoustic consultant."

17. *Softness:* Response is 5 dB down at 10 kHz; you can hear yourself think in the control room; the monitoring amps and/or speakers have blown up or melted; the session ended 3 hours ago, and you are wondering exactly where you are.

18. *Blurring:* Precursor to #1.

19. *Nasal:* The tweeter and bass unit have blown; you have patched in a compressor with pin 3 hot to your board, which has signal on pin 2.

20. *Glare:* Finishing a vocal overdub session at 10 a.m., and going out on to Sunset to get a beer.

21. *Eveness:* Tightly controlled sibilants ("even-ess"); reciprocal of #14.

I would like to thank the British engineer/producer who sent me these rather 'interesting" GE definitions; now if I can only get him to send me some straight questions . . . We will be putting the *real* Glossary together later on this year. There is still time to send in your definitions, so please share them with us.

From the definitions received so far, I think I can identify a trend. Many of the GE terms describe subtle differences in spectral balance (frequency reesponse). In some cases two or more words describe the same effect, depending upon the viewer's positive or negative perception of the result. To help quantify these terms, please include specific frequencies (guess), and antonyms if you know of them. Wouldn't it be great if we could line up all these terms to a frequency chart?

More Absolute Polarity

In the April issue I made the suggestion that, for archival purposes, we should attempt to maintain absolute polarity through our recording chain, based on the expectation that playback equipment will continue to improve, and polarity someday may be more audible than it is now. I would like to point out another way that polarity can get mixed up, and which isn't always obvious.

In 1975 the IEC issued Standard #268 part 12 covering three-conductor, XLR-type connectors. As defined by the standard:

Pin 1: *Shield*
Pin 2: *Signal* (+)
Pin 3: *Return* (-)

However, there exists another informal standard — sometimes called the "American" standard — which uses:

Pin 1: *Shield*
Pin 2: *Return* (-)
Pin 3: *Signal* (+)

There has not been a stampede among non-conforming manufacturers to comply with the IEC standard, since a polarity reversal at both the input and the output will tend to correct itself, and there is no problem interfacing balanced lines. It does get a bit messy with unbalanced lines though, and some quick cable rewiring often is required to get the system working. To the best of my knowledge, most microphones conform to the "pin 2 hot" standard, but polarity varies with mixers and other high-level gear. If the manufacturers will let me know which standard they are wiring to, I'll endeavor to print a list of who is doing what. In the meanwhile, check the small print and good luck.

Also on the subject of polarity, if you have added transformers to your system, note that there will be a dot on the secondary lead that is in phase with the dotted primary lead.

New Business: "Distortion Boxes"

In this month's column I would like to examine why perfectly sane people spend up to thousands of dollars to add distortion to their audio product (recordings, broadcasts, etc.).

Fuzz: Probably the largest (intentional) source of distortion is the guitar effect known as "fuzz" or "fuzz-tone." With a name like "fuzz" you wouldn't expect it to sound clean, and no surprises here; it doesn't. Instead, to my ears at least, it sounds rather like a severely clipped amplifier. The effect usually is created by clamping a signal with a diode, or by overdriving a pre-amplifier, which is how I expect the effect was created.

Why do people use it? Well, because the effect sounds good. Guitars/amps/speakers are not known for their wide frequency response. Processing the signal with a fuzz box adds many higher harmonics to give the signal more "edge" [1] to help cut through the mix, and "sustain" [2] to smooth the dynamics.

[1] ***Edge:*** *Refers to the balance between fundamental and higher frequency components of a given signal. More emphasis on the higher components will give a signal more edge. Sometimes referred to as "bite."*

[2] ***Sustain:*** *The ability to hold a note beyond it's normal decay. An instrument's natural sustain will be affected by playing style, body rigidity, and string termination. Sustain can be artificially extended by adding compression, or fuzz, and/or allowing the speaker to feedback to the strings.*

The fuzz effect I've been describing is usually found in under-$100 floor boxes. There is another type of distortion box invading the studio with a price tag in the thousands of dollars. As you might expect, these studio boxes don't sound like a clipped amplifier. What they do sound like is a bit difficult to explain, so I won't, at least not now.

For obvious reasons, the manufacturers of these "studio-quality" boxes do not call their effects "distortion."(Can't you just hear the producer saying to the mix engineer, "Could you put a little more distortion on the lead vocal!") Instead, they call these effects "exciters," not to be confused with the exciters found hooked up to radio station transmitters (they are *not* interchangeable).

How Do They Work, Why Do They Work, And Do They Work?

These studio "exciters" are much more sophisticated than their fuzz-box relatives. First of all the effect, when properly executed, is subtle. The output just sounds better. It is a serious paradox that adding distortion can make a signal sound cleaner, but that is exactly what happens.

Any classical definition of distortion would label the signals being introduced by these "exciters" as distortion. [The office *Webster's*, for example, defines distortion as: "falsified reproduction caused by a change in waveform of a signal" — *Ed.*] Our ears, however, do not always agree.

The reason these effects sound as good as they do is because they specifically create only the second harmonic or, in some cases, are adjustable between the second and third harmonic. Secondly, they do not operate on the whole spectrum, but rather a limited band of frequencies where signal fundamentals typically occur. This has the effect of reducing unpleasant intermodulation distortion by very high or very low frequencies, and also limits the created harmonics to frequencies where they already are naturally occurring. Signals rich with low-order harmonics are often described as "warm" [3].

[3] ***Warm, Warmth:*** *I don't know. Sounds like a lot of low-order distortion. Note: some tube designs noted for their warmth have overload characteristics that favor the lower harmonics.*

With the possible exception of the flute, which is almost pure (pianissimo 95% fundamental), most musical sounds are made up of complex combinations of fundamentals and overtones. On paper at least there is no difference between an instrument's first overtone and that fundamental's second harmonic. Properly introduced low-order harmonic distortion can sound quite musical; get a bit *too* heavy with the third harmonic and it starts sounding like distortion again.

While I don't see these boxes displacing proper recording technique, they do create interesting and useful effects. So next time you're mixing down and the vocal just isn't cutting through, you may want to ask the engineer to add a little distortion!

Reference: Patent # 4, 150, 253. Curt A. Knoppel; "Signal Distortion Circuit and Method of Use." April 17, 1979.

PS: If you are interested in figuring out how equipment works, patents make very interesting reading. Contrary to popular belief, the Patent System was not set up to enrich inventors. The government makes a deal with the individual inventor: in exchange for complete disclosure of the invention, he or she has protected use of it for a fixed period of time. The intention of this system is to discourage inventors from keepign their good ideas secret. As one good idea can often generate another, much can be learned from studying patents. If you are interested in how a device works, and there is a patent number printed on the back of the unit, or in the literature, you can send away to the Patent and Trademark Office, Washington, D.C. 20231; enclose \$1.00, and the number of the patent in which you are interested, and in a few weeks you will get the copy(s) in the mail.

Make sure you copy the patent number correctly, or you may end up with a patented doggie umbrella, or other such useful article! ■■■

INDUSTRY INVENTIVENESS IN THE EIGHTIES

A Profile of Film and Television Mixer BILL GAZECKI

by James Riordan

As sound recording techniques and applications continue to expand, it is predicted that a whole new horizon of related careers will materialize. Many of these new careers will be explored in future columns, but for this issue I'd like to look at what I consider to be a classic transition. At first sight, moving from recording music to recording all types of sound may not seem as though it would require a lot of new training. But when we consider the transition one makes in leaving the world of the record business for the world of television and film, we are talking about quite a transition indeed.

Bill Gazecki

Probably the most common "new" career for today's music engineer will lie in TV and film recording. Our interview this month is with Bill Gazecki, a man who after achieving a good deal of success in the music business as a producer and engineer saw that a future in film and televison sound could even be more fulfilling.

Gazecki is probably best known as associate producer of *The Rose* soundtrack, and as the engineer who helped Paul Rothchild put together *The Door's Greatest Hits*. He also has worked with such artists as Fleetwood Mac, Nick Gilder, Leo Sayer, Pure Prairie League, and Joe Cocker. Gazecki also designed and built a private recording studio for David Holman, which later was used to record the *Xanadu* soundtrack album.

Gazecki grew up in San Francisco during the Sixties, and by the age of 15 had secured the interesting job of selecting the music for an experimental encounter group specializing in psycho-

— *continued on page 127 . . .*

The 6120 is an original — not just a warmed-over copy of some other duplicator. It's brand new, and offers you more time-saving, quality features in one compact package than any other duplicator on the market today.

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16-to-1 copying speeds from reel or cassette. Reel modules run at either 60 or 120 ips and cassettes run at 30 ips, which means you can copy up to eleven one hour programs in less than two minutes!

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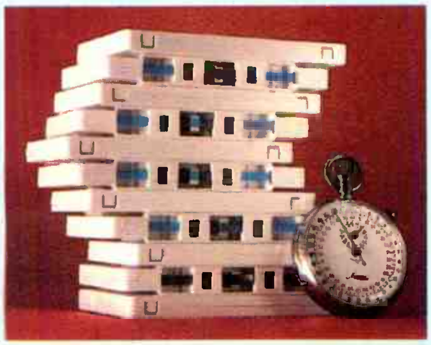
The new Telex 6120



You can produce eleven C-60 cassette tapes in less than two minutes!

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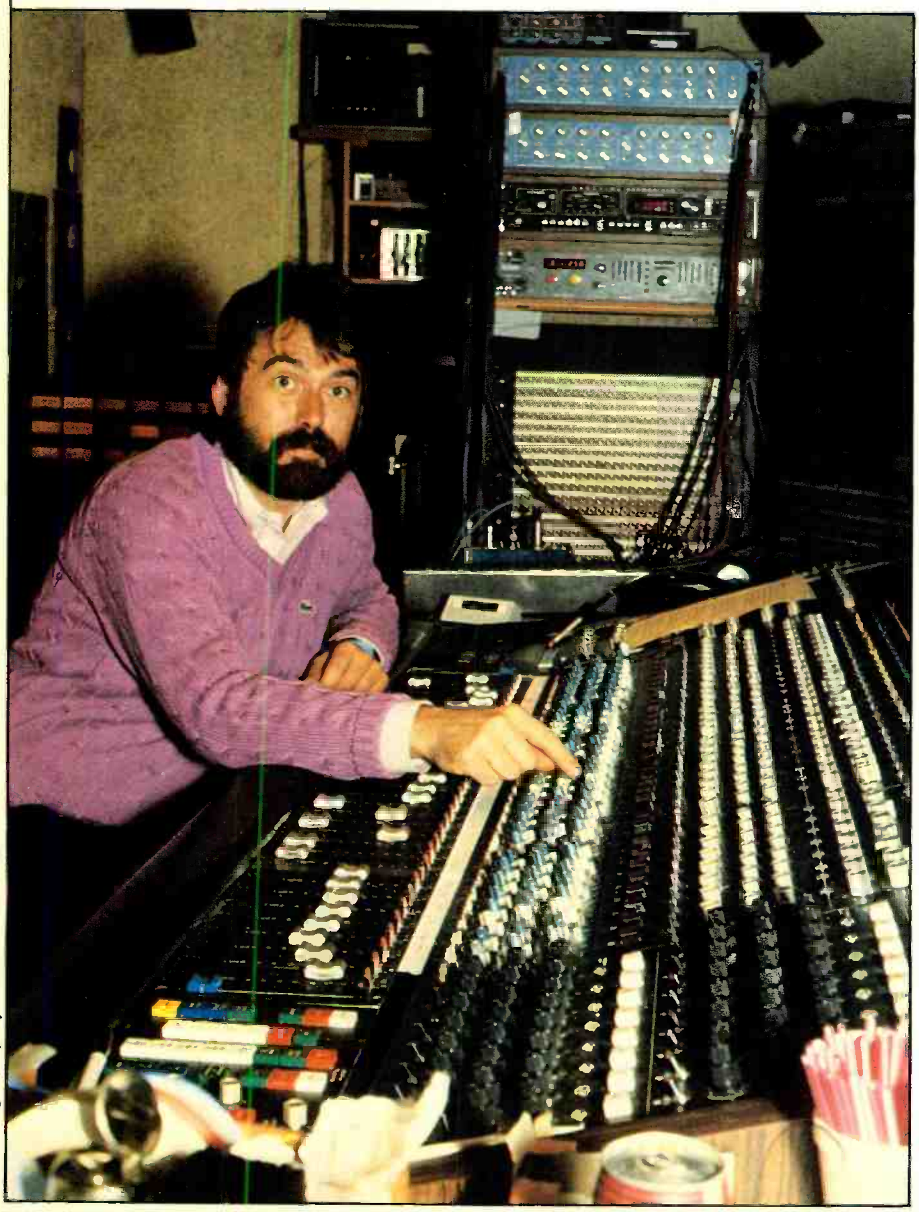
Peter McIan's career has included virtually every role in the recording industry: engineer, producer, studio manager, A&R staffer, songwriter, and recording artist; his reflections on today's music business are uniquely qualified. He first gained prominence with his solo album, *Playing Near the Edge*, which McIan wrote, arranged, and produced. A single culled from the album, "Solitaire," became a Top 40 hit in 1980. From there he went on to produce Franne Golde, and one of the songs they co-wrote, "Looking for a Stranger," is Pat Benatar's current hit. Following his success with Golde, McIan was introduced to Peter Karpin, who served as A&R man for CBS Australia. What had been planned as a five-week stay turned into an Aussie Odyssey. Karpin had just signed a band from Melbourne, Men At Work, and McIan dropped in at their first gig in Sydney. Inspired by their inventiveness, polished performance, and the unusual vocal qualities of Colin Hay, McIan and the band went into the studio and emerged six weeks later with *Business As Usual*. The album went multi-platinum worldwide, produced two number one singles, and is said to be the most successful record in Australian history. The group's second album, *Cargo*, also produced by McIan is selling like vinyl hotcakes. *R-e/p* caught up with this hard-working engineer/producer at Sunset Sound in Hollywood during sessions with an Australian band, Dear Enemy. What underlies his straightforward approach to recording and producing? What's it like to work down under? Pull up a fader and listen to the soft-spoken McIan tell his story.

PRODUCTION VIEWPOINT — U.S. & AUSTRALIA

PETER McIAN

Interviewed by David Gordon



All photography by KATHY COTTER

R-e/p (David Gordon): Do you feel that the producer's role is active like a director, or is he or she more of a babysitter?

Peter McIan: I think the producer is a "facilitator." He takes the audience viewpoint and expresses it to the band; then, by remaining as objective as possible, he creates a perspective. The directorial element is also involved. What I try to do in every song is to find the core of the material and bring that out, so that there's an atmosphere of creativity that supports whatever the lyric of the music is.

In terms of dealing with the personalities involved, I think part of production is trying to make it as easy as possible for those performers to *perform*. And, of course, it's important to spend a good deal of time trying to get the headphone mix right; trying to make it as pleasant and run as smoothly as possible in the studio. After all, the main object is to get a performance on tape, and that's what we're in there for. So there's a lot of jack-of-all-trades involved in producing. As far as the creative element is concerned, for me it's the best job in the world. You handle large creative elements as well as the problem-solving side of it, which is constantly changing. On each and every project there are different problems that arise, so it keeps you on your toes!

R-e/p (David Gordon): How did you feel at the time you left the States to work on sessions in Australia?

Peter McIan: The opportunity to go to Australia was terrific, but there was also a push from here. I was so bloody bored with American radio at the time. It seemed like all the records sounded the same — the "L.A. Sound" — which when it first started was innovative, but it had just been driven into the ground. A lot of people blame that on the session players and say that they're using the same musicians. But that's simply not the case. Those players, in most cases, are tremendous musicians, who are innovative themselves. The problem has been that the requirements of radio and production were to play it safe, and whenever you do that you automatically squash any innovation that's going to take place.

So I went to Australia and suddenly was exposed to music that I simply had *never* heard before, but which has since become successful here with bands like Duran Duran, The Human League,

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PRODUCTION VIEWPOINT

PETER McIAN

Stray Cats. There were a number of records by English and Australian bands that were being played regularly on the radio there, and having hits. In the last year that music has finally broken through in the United States, and to me it's a really exciting time here for new music. It's taken some doing on the part of the more innovative radio stations around the country, like KROQ in L.A., MTV, and those program directors courageous enough to put the numbers on the line and try something different — as opposed to the same old "chainsaw music," or the L.A. sound that has been going on for years.

R-e/p (David Gordon): What effect has the huge success of Men At Work had on your producing career, and your creativity?

Peter McIan: Well, to be honest, in a lot of ways I haven't had the chance to digest it! We've gone from the Men to other projects, and the obvious change is the ability to work more often. I now have the chance to do a record with a new band if I'm interested in them, whereas before it was a process of doing demos, and the usual chain of events that go into getting an A&R department to say "yes."

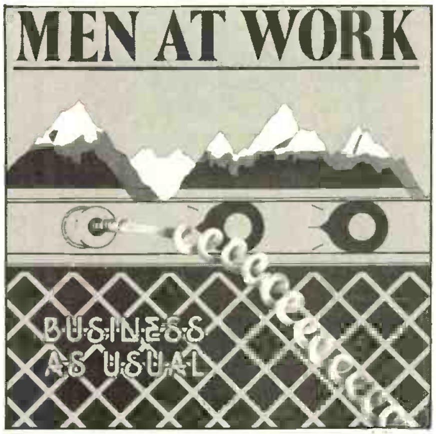
Now I have a bit more credibility, so that when I walk in the door there is at least an ear to hear whatever it is I'm playing. That has been the main change, and it's been the one that's most exciting because my principle interest has always been *new* bands. I find it much more interesting, much more exciting, to work with a new band than to do the 11th album by somebody who's established. This isn't a question of musical validity, it's just a personal preference.

To me it's really exciting to go through the pre-production period of rehearsal and so on, and then come out the other end with a band that has a little bit different identity, or a more solidified identity, and to hear the way the songs change. It's a *very* exciting process, and now I have the opportunity to do it more often.

R-e/p: It's our understanding that bands in Australia survive by their live gigs, and that they need to have a constant flow of fresh material in order to keep their audiences interested. Men At Work is reported to have had hundreds of songs in its repertoire. In your role as producer for the band did you take the best parts of various material and combine it, or was it a matter of focusing in on songs that you thought were strong, and then developing them?

PM: More the latter. It was a great position for a producer to be in, because they had a lot of *really* good songs. In fact, some of the songs on *Cargo* were written four years ago, although many of them are brand new. In working with them we picked what we thought were the 12 best from the standpoint of coherency in terms of an album. Then we went to work developing those songs and developing a sound, which I think is a key part of the pre-production process. It's *not* just the arranging and rearranging and rehearsing of songs. It also gave me an opportunity to hear the band, and begin to get ideas of what the record should sound like to support those songs. That's basically the way it came about. We probably rehearsed 20 songs, and chose 12 to record.

R-e/p: Many of the Men At Work songs start off with a very distinct introduction, almost a musical logo. Is that something that you put into the sound,



"The marketing importance of video is just beginning to come into its own in this country, and when it does I think that it's going to explode!"

while shaping them, or is it something that was already there which you brought out?

PM: I've always considered intoductions to be extremely important; they set up the mood of the song, and there's a lot you can do with an intro. You can do the old "sucker punch," where the intro is quiet and when the actual body of the song comes in you hit them over the head with it. Or you can establish the mood for the vocal with the intro.

In most cases, whenever we work on a song it's a cooperative effort and the intros are usually not the same as they were before we started working together. I've always considered them to be a *key* part of any song. I think to attract the listener's attention as early as possible is a pretty good idea, and with a strong intro that's what you do. In addition, we usually try to find a way to relate the intro to some portion of the song that occurs later in the instrumental section. But the main idea is to set an atmosphere for the vocal.

R-e/p: Isn't the flute riff on the song "Down Under" derived from an Australian folk song?

PM: That's right. "Kookaburra sits in the old gum tree," which is an Australian children's song. It was actually a bit of a musical joke that Greg [Ham] put in; in fact, in the video he's sitting in a gum tree while he plays it. The character of "Down Under" was Australian, and they felt that was very Australian. It was either that or "Waltzing Matilda," which wouldn't have fit!

R-e/p: Was the style of production and the way you set up Business As Usual *sessions the same as with all the artists you work with?*

PM: No, in fact it's very different. One of the things that I think is a case of *mis*production is when the producer stamps [his style on] every record he does. It's the *artist's* record, not the producer's. It's something that I feel very strongly about. The producer should listen to each band as a separate entity, and the way he records that band has *got* to be different. In the case of Men At Work, for instance, the echoes that we use are echoes that I thought were suited to them, to Colin Hay's voice in particular. On some of the mixes there are as many as 12 different kinds of echoes.

R-e/p: On the subject of using multiple echoes, how were they used on a particular song; say, "Who Can it Be Now?"

PM: "Who Can it Be Now" was a simpler mix, but on that one there's a slap echo on the voice; there's a gated echo on the snare; there's a delayed chamber; and a straight chamber set at different decay times.

My personal approach is echo mixing, more than level mixing. What I try and do is visualize a room in which all these people are standing and playing, and then to create as much as possible a balance between the instruments by using echoes that are further or closer apart. "Down Under" was a bit more complicated. There was a phase chamber, a delay chamber, straight chamber, a Roland 555 [Space Echo], and a short echo on the snare.

For the mix we used EMT 140 [echo plate] chambers, and one EMT 250 [digital reverb]. We used a long decay time on the chamber for the drums. On the snare we used a delayed chamber with a gated echo return. On the backing vocals we used a phased chamber; on the synth we used a short echo; and on the vocal the Roland 555, an analog tape, "space echo-type" device, which we split off to stereo using a delay line. All the echoes are fairly subtle in the mix so that they wouldn't sound like discrete echoes, but instead as if they were all *part* of the same environment.

R-e/p: The sound of the Men At Work albums is different from prevalent production styles these days. There is a presence of the band up front and effects which wash back, rather than the whole

RECORDING IN AUSTRALIA

In Peter Mclan's experience with Men At Work and Mondo Rock, were there any main differences between recording in the States, and recording in Australia, we asked.

"The principle difference is maintenance and the equipment that is available in the studio," he offers. "Australia is at the end of a 12,000 mile pipeline, but there are several state-of-the-art studios. Paradise in Sydney is an excellent facility, and has a Sierra/Eastlake room. The guy who runs it, Billy Field, is a major star himself in Australia, and he's extremely conscientious about the operation of the studio. AAV in Melbourne is taken from Westlake drawings, and modified; it also is state-of-the-art. The problem, however, is that if something breaks, spares are hard to come by. You can really be up the creek if your pitch shifter, for example, goes south — getting another one can be difficult.

"I did the Men At Work album at a little studio in Melbourne called Richmond Recorders, and the console it was recorded on was a little MCI JH-400B with fixed-position EQ. I begged, borrowed, and stole every piece of outboard gear I could get my hands on in the city. The showrooms were empty when I was mixing that record, and we own a lot of favors to the various equipment people there!

"But the biggest difference is maintenance. The really good studios there are booked solidly, and there isn't an attitude of preventative maintenance the way there is here. Because the studios are so busy, to go down and do preventative maintenance isn't a realistic alternative for them. Maintenance there takes the shape of fixing things, rather than keeping them from breaking.

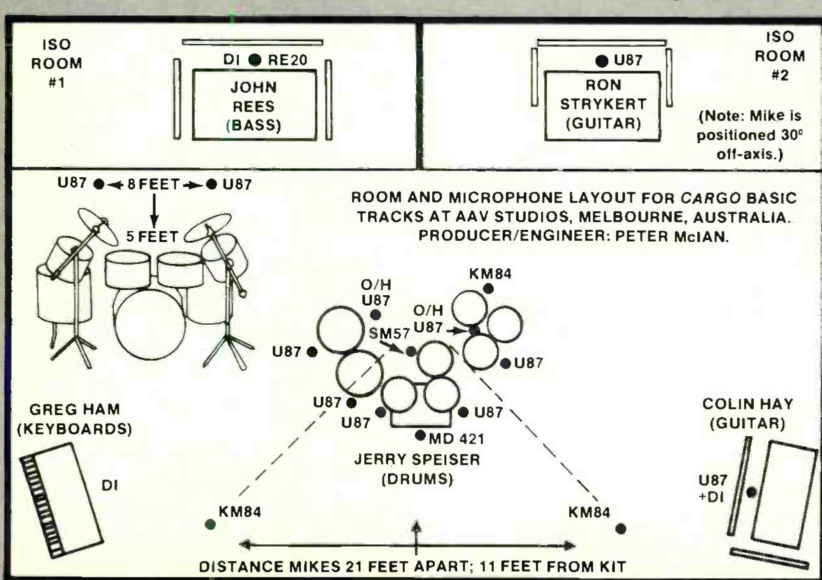
"The other thing that's a real factor is that the community of engineers and maintenance people is quite small. It's a country of 14 million people, which is equivalent to the population of New York City. So the interaction within the engineering community is limited relative to the United States or England, where there are skedillions of engineers and maintenance people around and, as new ideas emerge, those new ideas are thoroughly hammered home.

"In Australia it's oftentimes a trial and error process. They simply aren't allowed the luxury of experimenting. The studio's going all the time so they can't take a chance on biasing a machine a different way to see how it works out. They have to pretty much stick with what works."

Is the new popularity of Australian music helping to cause growth in the recording industry?

"Yes, it is, very definitely," Mclan replies. "But when you consider that the Men At Work album was done for about US $30,000, compared to what would be a normal budget in the United States, studio budgets are related to population size. If your market is only 14 million versus 250 million, then the amount of money you can spend recording a record is limited. I'd say the average budget in Australia had been 20 to 25 thousand dollars to do an entire album. That means that studios obviously haven't been able to expand their facilities because there just aren't that many dollars coming in the door. You don't have the luxury of having 14 Harmonizers and 22 DD1s, and so on.

"That's changing because more and more Australian albums are being released overseas, and there's a much broader acceptance. Traditionally, American record companies have been extremely chauvinist about what kind of records they accepted. One of the big problems that acts in Australia faced was the fact that international A&R departments in the United States didn't have a lot of clout with the domestic A&R departments. A lot of good records just never saw the light of day. Now there's a much broader acceptance at record companies of music that comes from places other than the U.S. and England. This means



production being enveloped in one sound . . .

PM: That's all echo mixing. Clarity is something I really shoot for. I figure if a player is playing something that is worth hearing, it's important to be conscious of the interplay between all of the various parts. A song is a lot like a short film, and you can create an ambience and an atmosphere with echoes the same way you would with [lens] filters while shooting a film.

I tend to use echo as color, and try and create an atmosphere that's suited to the song being performed. We start from scratch and set up the echoes differently in each and every case. One of the things that I've never been a big fan of is the "wash" of echo that blurs everything into one mass. It's more time consuming in the mix, but I try to end up with an *atmosphere*, as opposed to just a song.

R-e/p: How do you set up for the basic tracks?

PM: I try and record as complete a song as I can. I don't like to do a lot of overdubbing. In that respect, going to Australia was a real education for me, because it gave me the opportunity to work with bands who play live constantly. It's very important to them, and it became very important to me, that they be able to translate what's on record in a live performance. To do a whole lot of overdubbing and create a song that way, or create the impact that way, is self-defeating where they're concerned, because they hit the stage and can't duplicate it. So when we go in to record, what we try and do is set up so that as close to a finished song as possible is going to come out of the basic tracks. Then the overdubbing is left for things like final vocals, guitar solos, those sorts of things. There really isn't a lot of overdubbing on those records, and in the case of Dear Enemy, the band I'm working with now, the same is true. We're just about finished with the basic tracks, and the overdubs will consist of the percussion part, lead vocals, backing vocals, and guitar solos. I try and set it up so that we can end up with a finished song.

R-e/p: Do you isolate the drums while tracking basics?

PM: No. I tend to like to use the drums in as big a room as I can, and will try and get the amps away from the drums because of the overhead miking technique I use. When I was in Australia I developed a miking technique for drums that is flexible and fairly quick. I can go into almost any room and set it up using a triangle miking method combined with close miking. Beyond that I can collapse or expand those triangles as the size of the room allows.

R-e/p: Which microphones do you use on the kit?

PM: Over the drums and behind the drummer I use two [Neumann] 87s, so

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PRODUCTION VIEWPOINT
PETER McIAN

that I catch the toms and cymbals. On the tom-toms themselves I use 87s again, and they're miked usually two or three inches off the drum head. I never dampen drums; I always leave them wide open. There's a certain kind of drum head which I use all the time, known as Ambassadors. The only damping I ever use is on the kick drum, and I use a bit on the snare if there's really a horrible ring. Then in the mix, I gate the toms, because I figure if you're damping the drums and you spend all day going for the perfect tom, you end up with a very tired drummer. The object of the exercise is not to get the perfect tom, but to get a great performance.

With that in mind, I try to develop techniques that are quick and I know are going to get me a certain result, and then in the mix I can make alterations as need be by using gates, echoes, EQ, and so on. By using this technique I get a certain consistency that allows the drummer to make the difference. *His* playing makes the difference.

I know the kit is going to sound pretty good, and there's also a tuning approach that I adopted where I generally tune the toms to the key of the song. I tune them fairly high, and then tune the bottom heads so that they don't resonate in sympathy with the top head. I try to eliminate that as much as possible, so that we get the bang out of the toms without a whole lot of over-ring, because there's no damping at all.

R-e/p: Snare is really prominent in your productions. How do you mike the snare drum?

PM: A [Shure] SM-57 on the top, and then a [Neuman] KM84 on the bottom, if I have one available. The problem is that in Australia, as in a lot of studios, rather than adopting a wide range of microphones a facility will have its own style, and may favor a particular kind of microphone. So sometimes I've used Beyers because the studio will have bought a lot of them. Knowing what your chosen microphones do, you try and compensate. But generally it's mostly Neumanns and a Shure 57 on the snare.

R-e/p: And on the kick drum?

PM: I use a [Sennheiser] 421 on the kick, which I'm told is not the mike to use. But, then again, I usually do things in a fairly unorthodox way! When I record I horrify second engineers because I saturate tape a lot — deliberately — for an effect on certain instruments.

R-e/p: How do you mike guitar amplifiers?

PM: I've used all sorts of mikes on guitars. For power stuff I tend to use a

dynamic. For quieter things, for cleaner [sounds], I'll usually use an [AKG] 414 or an 87. I set it usually 30 degrees off axis, and that's pretty much it.

R-e/p: Do you print echo while tracking basics?

PM: No, unless it's a situation where the guitarist is using echo through his amp. Then it gets printed. But generally, I use [the bands' echo unit] in the mix, and then take it off the amp, so the "air" is actually echoed. A bit of the ambience of the guitar is echoed, rather than trying to echo in the amp itself.

R-e/p: How do you record electric keyboards?

PM: I usually take the keyboards direct, unless we're looking for an effect. In that case we'll go through an amp. But, generally speaking, they come in direct. I've found that I don't like to use a lot of EQ on keyboards, so I just try and get rid of the garbage at the extreme top-end, and that sort of thing.

In every song there is usually a key instrument that is the energy source in the song. In some cases it might be the drums; in some cases it's the guitar or keyboard part, and in others the vocal. What I do in terms of miking technique — and in terms of whether to use a clean guitar sound, a distorted sound, or whatever — is in rehearsal to try and find whatever that energy center is, and build the rest of the song around it. For instance, in that triangle miking technique that I was talking about, I have two room mikes and two overheads that are both in triangles in different dimensions from the snare. If it's a song that's a ballad, or slower-paced, I'll tend to collapse the triangle a little bit. If it's something that's a real rock 'n' roll song, I expand them, but the distance remains proportional as much as possible. The size of the triangle will change depending on the song, the amount of room ambience, and all that sort of stuff. It's the same with guitars. For some guitars, I might use a close mike and a distant mike as well, and mix the two together. And then for other songs I might use only the close mike.

R-e/p: How about vocals? Is there anything special you might use to capture the vocal quality you're after?

PM: I usually use two microphones on

RECORDING IN AUSTRALIA ... continued —

that studios in Australia are able to grow, and bands are able to spend a little more time doing a record, instead of having to mix an entire album in two or three days.

"It's a real testament to the ability of Australian producers that they have been able to come out with really good sounding records with that kind of time pressure."

What advice would he offer to a producer or engineer planning to record a project in Australia, we wondered?

"First of all," he suggests, "if you want any exotic gear, like valve microphones or rare equalizers or that sort of thing, you'd better plan on bringing it in yourself. It's getting better there, but it's still at the end of that 12,000-mile pipeline.

"I'd also say that if you want your machine biased in a particular way, or if you want it aligned a certain way, you'd better be able to do it yourself. In each studio there is usually a definite way of doing things; if you want it done differently you should plan on doing it yourself. Other than that, for the most part, they have state-of-the-art studios.

"I would suggest taking a good reference monitor, something that you know, like a JBL 4311. I carry my own monitors with me wherever I go — the new Westlake BBSM-6s. I also carry my own outboard equalizers, and two dbx Over-Easy compressor/limiters that I use a lot. They're a real gentle compressor, and they aren't terribly popular with studios [in Australia], so I carry them with me. I also carry a couple of Roland 555s.

"So if you have preferences for certain things you should bring them in with you, and that you also have a set of monitors that you're familiar with to judge the monitors in the room by. There can be pretty radical differences from one studio to the next."

The band you're working with now, Dear Enemy, is an Australian group that you plan to record at Sunset Sound in Hollywood. Why here rather than in Australia?

"Because the studios were booked up in Australia," McIan confides. "We had time reserved, but there were some administrative delays and we lost our time.

"Australia is a country that has looked at the world's music through a telescope from 12,000 miles away. The brightest elements from various countries become the influences in Australia and Dear Enemy really typifies that. There's a lot of the English quality to them in terms of the use of synthesizers, and then there's a lot of the American influence in terms of the vocals. The lead singer, Ron Martini, is an extremely powerful singer, and there's five-part backing vocals — they all sing. At the same time, they're uniquely Australian. One of the delightful things about Australian music is caused by the live situation there. The bands develop a unique flavor that only playing live regularly can give you, and the same is certainly true of Dear Enemy. ■■■

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PRODUCTION VIEWPOINT
PETER McIAN

vocals. I use a 414 set back three or four feet, and then in front, depending on the singer, I'll use either an 89 or a tube 47; those are the two I use the most. In one case, for a distant mike I've used a [Crown] PZM instead of a 414.

Basically, what I try and do is get the warmth out of the close mike, and get the breath in the top-end of the distant mike. In Colin Hay's case that was particularly important, because he's got that wonderful throaty quality.

R-e/p: At times his voice almost seems like a reed instrument.

PM: Yes . . . and it was difficult to capture. We had to work quite a lot in terms of finding the right miking technique for him. On "Who Can It Be Now" I used a PZM mounted on a mike stand, and an 89. On the rest of the album, and also on *Cargo*, I used a 414 and an 89, and we had to fool around with distance because of phase cancellation.

The two mikes were equalized completely differently. The distant mike was EQ'd for the top end of his voice —for the "air" — and the 89 for the body of his voice. The two were mixed differently for each song, depending on how loudly he was singing, and the compression we had to use. But in every case the 89 was the dominant of the two microphones.

Colin [Hay] is a terrifically creative singer, but he's also a very professional singer and is willing to put in the time to find the right approach to get the sound of his voice. It's a bit of a problem voice to record — he's a very powerful singer. It seemed to me that if I used just the one mike I'd get the point of the voice, but I wouldn't get that wonderful "reedy" quality. That's why we went to the two mikes. And, of course, those are mixed down to one track and recorded.

R-e/p: What is your general approach to the mixing process?

PM: Generally, the night before I set up the mix I do a rough. Then the next day I come in and take it after I've had a chance to sleep on a particular mix, and fine tune it. I usually get a mix a day. Most of the time is spent in setting up the multiple echoes, and how we're going to use them.

To me, the mix is like the final edit of a film. I think it has a great deal to do with what kind of atmosphere you create for the song. We spend a lot of time getting that atmosphere, and I usually map out the mix well in advance. I know what I'm going to use as a starting point, and then from there we make changes as I listen to it. But there's definitely a concept of what the mix is going to be before we start on it.

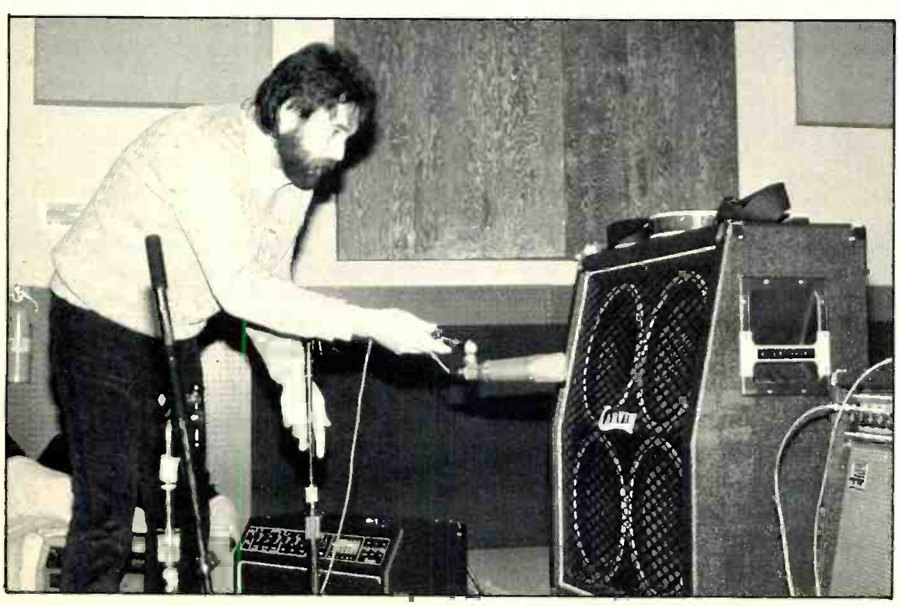
R-e/p: There seems to be a darker mood to the sound on the second album, Cargo. *Can you imagine "overkill" being a single that could have broken the group into the charts, the way that "Who Can It Be Now" was a huge hit?*

PM: No, not the way the other songs did. I can imagine it being a record that got a lot of airplay. "Overkill," to me, is a really effective song, and I think it's a terrific piece of songwriting.

R-e/p: What I meant was that as a group becomes more successful they have the opportunity to stretch out a bit, in terms of songwriting and possibly production?

PM: Exactly. The approach we adopted on the second album was to sit down before we started, have a series of discussions, and listen to material. What we felt was that a second album is an opportunity to get to know the band better. If you look at the first album as a first impression, then just like meeting a person — as you get to know them you find other sides to them — that's what we wanted to try and do with the second album. The approach was to place less emphasis on the more accessible things, and place a bit more emphasis on songs that, for lack of another word, have

Peter Mclan with associate engineer Paul Ray



more "meat on the bones." An act that's successful has the ability to take that little extra bit of privilege.

R-e/p: In terms of production, the second album seems to be more "layered," to have a less sparse sound. Did you have more time to spend on the second album, and which resulted in a different sound?

PM: Actually, the second album took about as much time as the first; there might have been two or three days difference. Believe it or not, there aren't that many more overdubs on the second album.

The parts themselves are more complex; there's more interplay between the various instruments, and we deliberately went for a "warmer-sounding" record, because of the music itself. The music has a bit of a richer quality to it than the first one did. It's a bit moodier, and a bit darker. What we attempted to do was to support that music with the sound of the record.

Generally speaking, we used less of the slap echo on Colin's voice for the second album. We tried to make the coloration of the echoes a little richer, because we felt it suited the mood of the music. We used longer decay times, and there were more delayed echoes. The mid-range is slightly less prominent in the second album because we wanted a "translucent" quality to the mix. If the first album has hard edges, the second album *softened* those edges. Also, we wanted the second album to be more "atmospheric." Consequently, the echo usage on the second album is fairly complex.

R-e/p: Can we pick one song to serve as an example of that difference? Maybe "Overkill"?

PM: "Overkill" is a good example. I think there are 12 different kinds of echo on that song. The sax has a slap on it that's a different time [interval] than

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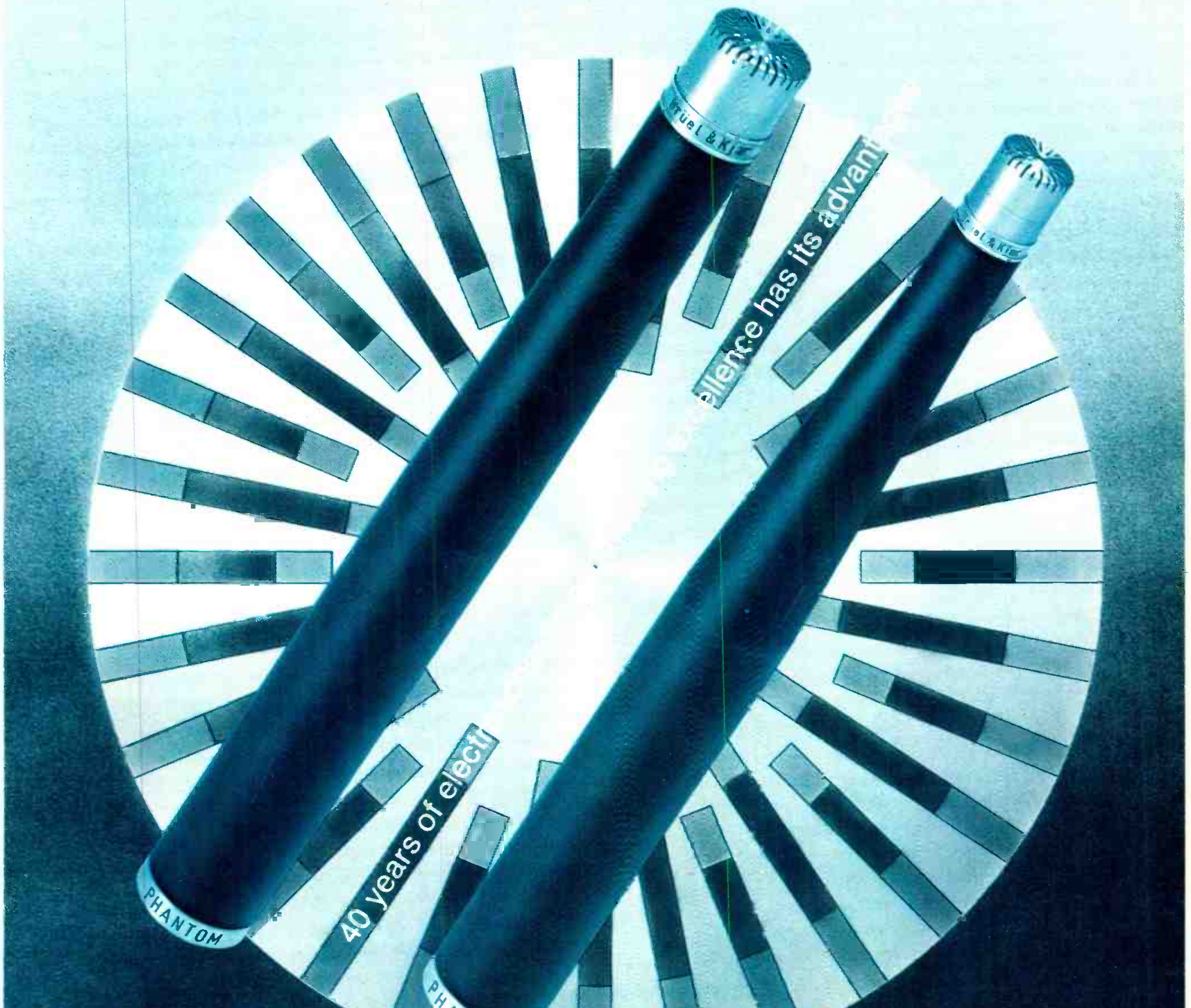

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PETER McIAN

the vocal slap. The drums have both a delayed and a straight echo on them, as well as a gated snare echo. There's a phased echo on the [Roland] vocoder which is used in certain places in the bridge. Actually, the vocoder is used as an echo

R-e/p: You have an unusual way of setting up the console during a session. Could you elaborate on that?

PM: It's a bit easier on a Harrison than on an API, which is the one I am using here [at Sunset Sound, Hollywood]. We were talking earlier about creating an atmosphere for the performers to perform in. Having been an artist for a long time, I know that a *big* component of creating that atmosphere is getting the right headphone mix, so that it sounds like a record in the cans when you're cutting a track. The way that most monitor sections are set up is that the feed to the cans is not EQ-able. Basically you're getting what's going on tape with the same EQ. Well, a lot of the time that isn't suitable for headphones; it ends up sounding muddy and the clarity isn't there.

What I do is I mult [parallel] the [tape machine] inputs to other channels as returns, and then send it out to the cans with different EQ than what goes on tape. We EQ a return so that in the headphones they get an EQ that's more suitable for performing. For instance, we can use things like Harmonizers in the headphones so that the musician can hear the [pitch shifter] guitar. And we can get a fairly suitable EQ, so that the bass player doesn't hear just a wall of mush at the bottom; he can actually hear his pick. Guitar players can hear what they're doing, and their fingerings, and the vocalist can hear what he's doing.

One of the few complaints about [some consoles] is that the monitor side is the weak link in the chain. You're getting what's going on tape, with a little echo on it. In some cases panning is not as accurate as it is on the channel returns, and so on.

R-e/p: Did you develop this technique yourself?

PM: Yes, in Australia. Working in Australia gave me an opportunity to grow as an engineer. I've always been fairly unorthodox and there are a lot of things that I've always felt would work in theory, but have never been able to try. When I went to Australia I was suddenly confronted with a situation where I had to do unorthodox things to get around certain limitations that were presented to me. The first Men At Work album was done at Richmond Recorders, which is a lovely little studio but very limited in terms of the board. Their console sounds really good, but the EQ is very limited. We had to find ways around that. It gave me the opportunity to develop a style of engineering that was flexible, but with a good bottom line. So I knew that when I walked into any room I could use the same setup as a starting point, and then vary it and know that at least I would get consistent results. I was able to hammer out some formulas while working in Australia.

R-e/p: You still engineer your projects, but you also work with someone else these days.

PM: I work with Paul Ray, who is an excellent engineer and a first himself. We work cooperatively, and my main function is as producer. Engineering is a tool to get the sound that I want to hear on tape. It gives me the luxury of going

"I tend to use echo as 'color', and try and create an atmosphere that's suited to the song. It's more time consuming, but I try to end up with an 'atmosphere,' as opposed to just a song."

one-on-one with a musician, where it can be just me in the control room and him in the studio. I find that's particularly useful with vocals.

I first met Paul while working on my solo album, when Mick Gauzauski was engineering. Mick now works with Earth, Wind, & Fire. He is a brilliant engineer and the guy I really learned from. I not only learned the manipulations of the console, but also ear training. I learned to hear things that I took to be background noise before! It's a blessing and a curse, because now if I listen to my car radio [sometimes] the static drives me nuts, whereas before it never bothered me.

A lot of it is ear training; a combination of the musical side versus the engineering side. I think a lot of producers don't know the instruments they're dealing with. They come into the studio and they know their *own* instrument, which is the *console*, and they're brilliant at that. But they don't know how a drum works, or how a guitar works — which fingering is the right fingering, and which position is the right position for a guitar player to get the voicing that seems to suit the song. These are things that should be really *basic*. It doesn't mean that you have to play the instrument; it just means that you have to have an understanding of how the instrument works, just as you have an understanding of the microphone and the console.

R-e/p: How did you acquire that musical understanding?

PM: Well, I was a musician first — keyboards and guitar. And I've been a songwriter for years, an A&R director, and I had a 21-piece rock band — all sorts of various things. I've been very lucky to have kicked around the industry for a long time picking up those things!

R-e/p: How is it different being the guy behind the scenes, so to speak, rather than the artist who's got his or her name out front?

PM: I can relax! I will do another solo album but it'll be much more of a record, as opposed to building an act to go on the road. I love writing, and I love being in the studio. I've got the best job in the world, and I much prefer being behind the scenes. I prefer working with bands because it's the collaborative effort that I find *exciting*. After the first time that I walked into a studio I didn't want to leave. I became the archetypical "studio rat," and at one point I also managed a studio — that gave me the chance to see how the business side of it worked.

R-e/p: What is your background as a musician?

PM: When you start out you're doing classical things, and once you develop the technique you have to forget what the boundaries were. When I had a big band I taught myself to read and write music. I would get a classical score out of the library, listen to the record, follow each instrument all the way through, and learn that way. Then I started writing for orchestras, and a man by the name of Carrol Hurak, who had been a second generation protege of Antonin Dvorak, took me on as a composition student. I was writing all sorts of bizarre things and one day Mr. Hurak, who was 90 at the time, asked me why I was writing things like parallel ninths, which are a no-no. I said I liked the way they sounded and he said, "Good, that's the *best* reason to write them." It was almost like having a sanction to break the rules! Sometimes ignorance is bliss, and if you don't know what the rules are in the first place, they're a lot easier to break.

R-e/p: What aspirations do you have now as a composer?

PM: I would like to work with synthesizers. I just bought an instrument called The Wave, which is a second-generation digital synthesizer. It has all the facility of a [Sequential Circuits] Prophet and

PRODUCTION VIEWPOINT

PETER McIAN

the sound quality is something like a Fairlight, or an [E-mu Systems] Emulator. It's very easy to use. When this record's done I'm afraid I'm going to become the mad scientist and lock myself away and just play with it!

R-e/p: Are you using it on the Dear Enemy sessions at Sunset Sound?

PM: Yes, I just got it about two weeks ago. It's the synth that Thomas Dolby ["She Blinded Me with Science"] has been using, and his work is certainly a high recommendation for almost any synthesizer.

R-e/p: Have you had any experience with digital recording?

PM: Well, we did the first rock 'n' roll tracks ever done on a prototype 3M 32-track digital machine for my solo record at Westlake (Studios, Los Angeles). The bottom end was amazing — it sounded tremendous, but the top end wasn't particularly good. I haven't used any of the newer digital machines because I tend to like analog tape.

R-e/p: What about the deterioration of analog master tapes during storage?

PM: That is the one *big* problem. The ideal situation — and it's something that I hope to do in the not too distant future — is to lock up a digital machine and an analog machine for recording. Certain things are better suited to analog, like cymbals and material with a lot of top-end information, or things where you want intermodulation to soften the sound. For instruments where you want punch and clarity, the digital machine is the answer. To be able to lock the two of them together, [analog and digital], which you can do easily with SMPTE timecode, would be the way to go, and then to mix. I'm now mixing to half-inch at 30 IPS and then to transfer to digital you would get the warmth of analog tape. By transferring to digital you avoid the deterioration problem, and you have no generation loss.

McIan with members of the band Dear Enemy, during remix sessions.

R-e/p: It must be great to have access to all of this recording technology now . . .

PM: You bet it is. There are a lot of ideas I've had in the past, and now I have the opportunity to try them out. Obviously, the financial side of things is terrific but, for me, the most exciting part is the fact that I get to work with people that I enjoy working with and I get to try ideas and continually expand on them in a way that doesn't jeopardize the quality of the record. I have the freedom to try out new noises and see what's going to work.

R-e/p: Were you involved with the promotional videos for Men At Work?

PM: No, not at all, they generated those themselves. That's another one of their talents, and I think we're just beginning to see the tip of the iceberg. I think "It's a Mistake" is the best video I've seen in a long time. Men At Work is one of the few bands capable of generating videos exclusive of the music. They played an instrumental piece for me when we were doing *Cargo*, called "Picnic in the Park." It's not a hit record, but a terrifically fun piece of music and they had a video script for it. They're one of the few bands that can do music specifically for video that would be in and of itself an entertainment. It's a field that I'm sure they're going to get into in the future.

R-e/p: Do you have any plans to become further involved in the increasing marriage of audio and video?

PM: Again, Australia was a tremendous eye-opener for me. Video in Australia is *key* to the success of a record. As an A&R guy I was fascinated by the opportunities that television affords for music, particularly where new acts are concerned. My background was originally in theater, where I acted in summer stock and did some directing. I'm starting to read the books and getting involved in video. I don't know how active I will be in actually generating the videos, but I certainly want to be involved in the coordination of the videos with the records I'm producing.

The marketing importance of video is just beginning to come into its own in this country, and when it does I think it's going to explode. There's a limited market at this point for selling videos to consumers because the "burn-out factor" is so great. There aren't a lot of people around who can generate a 60-minute video that is going to be interesting after two viewings. Also, the new technologies that are developing are going to bring down the cost of video.

I'm waiting with baited breath for the Compact Disc. The idea of a needle scratching across a piece of plastic is pretty bloody primitive! When the CD comes along and penetrates [the consumer market] it's going to be delightful. I'm sure the same thing is going to happen in video. We're going to end up with new technologies that make it more realistic and bring the cost down. So I hope to be involved in that when it happens.

The production team of McIan and associate engineer Paul Ray.

R-e/p: You also were involved as producer with another Australian band, Mondo Rock?

PM: Yes, Mondo is a band that is fronted by a fellow by the name of Ross Wilson, who is a legend in Australia. He was the producer of probably the most successful band in Australian history before Men At Work, a band called The Skyhooks, and as an artist Ross was the lead singer and key writer for Daddy Cool, a hugely successful band. A song he wrote called "Eagle Rock" is sort of *the* Australian classic rock 'n' roll single. They were a delightful bunch of guys to work with, and we did an album called *Nouveau Mondo* together that did quite well for them. From what I understand it's going to be released here at some point in the near future.

R-e/p: Will you be producing the third Men At Work album?

PM: Well, it's a year away, so we'll see what happens. As people we get along real well, and they're an amazing group to work with. I can honestly say that I can't imagine enjoying sessions more than I did with them. They're funny, witty guys, and being in the same room with them is a real pleasure.

R-e/p: Do you think that record companies are going to start sticking your name on albums, saying: "From the Producer of Men At Work"?

PM: I hope not. No, they won't — I can tell you they definitely will *not*. That's exactly what I was talking about before. The record that I make with an artist is the artist's record, not *my* record. I don't think I'm doing my job if there's a "McIan stamp" on the record. If they start stickering them I think I'm in big trouble because that tells me that I didn't do my job very well, and the record isn't as good as it should be. ■■■


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APPLICATIONS OF THE CENTRAL ARRAY DESIGN PROGRAM FOR COMPUTER-ASSISTED OPTIMIZATION OF LOUDSPEAKER SYSTEMS

by John Eargle, JBL, Inc.

The design of central loudspeaker arrays or clusters for sound reinforcement often has not been a direct process, and much time needs to be spent in the field re-aiming components after the system has been installed. In recent years, some manufacturers and sound contractors have developed their own mapping schemes, in which transparent overlays of horn directional contours can be placed over a mapping of the seating plane, thus indicating the horn azimuth, rotation, and elevation angles for best coverage. Two-dimensional mapping of three-dimensional space always will result in some degree of error, however, and the user of such systems must allow for these inaccuracies.

During 1982, JBL began the development of an ambitious microcomputer-based program for solving such problems directly; the Central Array Design Program (CADP) is the result of this work. Because of its graphics capability, the IBM Personal Computer was selected as the computer system for program development. The standard IBM Personal Computer with 96 kilobytes of memory and two disc drives is required to run CADP, along with a color monitor (preferably high-resolution), and a dot matrix printer for hard-copy printout.

In using CADP, the Cartesian coordinates of a seating area are entered into the computer, along with the location and angular orientation of one or more loudspeakers. In the first part of the display, the user is able to observe the relative sound pressure levels existing on the seating area as determined by inverse square losses, and the directional characteristics of the loudspeaker array. Each point displayed is the result of an exact calculation.

When appropriate acoustical data has been entered, the user can observe direct/reverberant ratios, as well as estimated intelligibility at all displayed points on the seating area. The mechanical design portion of the program produces front, top, and side views of the loudspeaker array, with an adjustable scale.

All calculations made in the program are based on standard equations which have traditionally been used in sound system design; more on this later.

Program Format

The CADP program is contained on one single-sided, 5¼-inch floppy diskette, and has been compiled from a PC.DOS BASIC source code for speed of operation. A data diskette also is provided, and contains polar and mechanical information on various JBL HF and LF systems that are commonly used in system design. Polar data on each loudspeaker unit is read in spherical coordinates every 10 degrees around the entire sphere surrounding the loudspeaker, as measured in free space. [JBL currently utilize three, rather than just a single, data diskettes to store HF polar data, LF polar data, and drawing files, respectively — *Ed.*]

There also is space on the data diskettes for storing job information, and it has been recommended that the user make multiple copies of data diskettes for future job use. Ideally, there should be one data diskette dedicated to each design project. In addition, the data diskettes can be used for entering and storing directional and mechanical data on any non-JBL HF or LF elements the designer may want to use.

CADP has been formatted to be easy to learn. Being a "menu-driven" program, the user simply responds to an array of options contained in a hierarchy of menus. Figure 1 shows the basic menu layout for the program, carried down to the second level, and provides a broad idea of the program's structure.

At this point, let's consider the content of each second-level menu, and detail the various equations used by the program.

Room Menu

This menu asks for all data pertinent to the room in which the loudspeaker array is to be located. The following acoustical data is requested by the program (the user can select to work in either feet or meters):

1. Volume of room.
2. Surface Area of room.
3. Average Absorption Coefficient of surfaces.
4. Reverberation Time of room.

Only one of the last two items listed above is required — either absorption coefficient or RT60; the program will calculate the remaining one. The Norris-Eyring reverberation time equation is used in the following forms:

$$T = (0.16V)/(-S \times \ln(1 - \bar{a}))$$

$$\bar{a} = 1 - \exp(-0.16V/TS)$$

If the required acoustical data is not readily available, the Room Module part of the Room Menu can be used to calculate the acoustical quantities. The Room

— continued on page 44 . . .

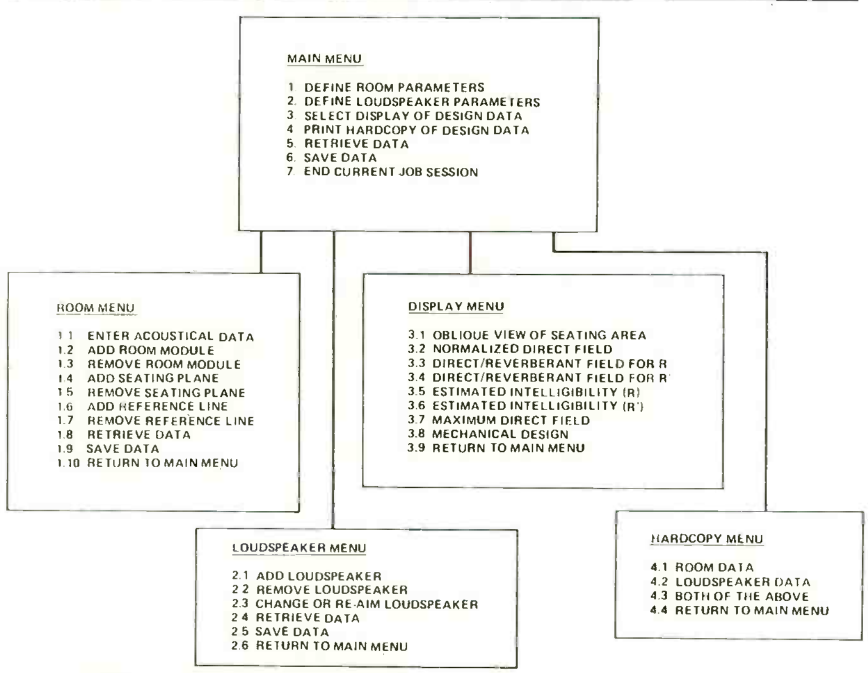
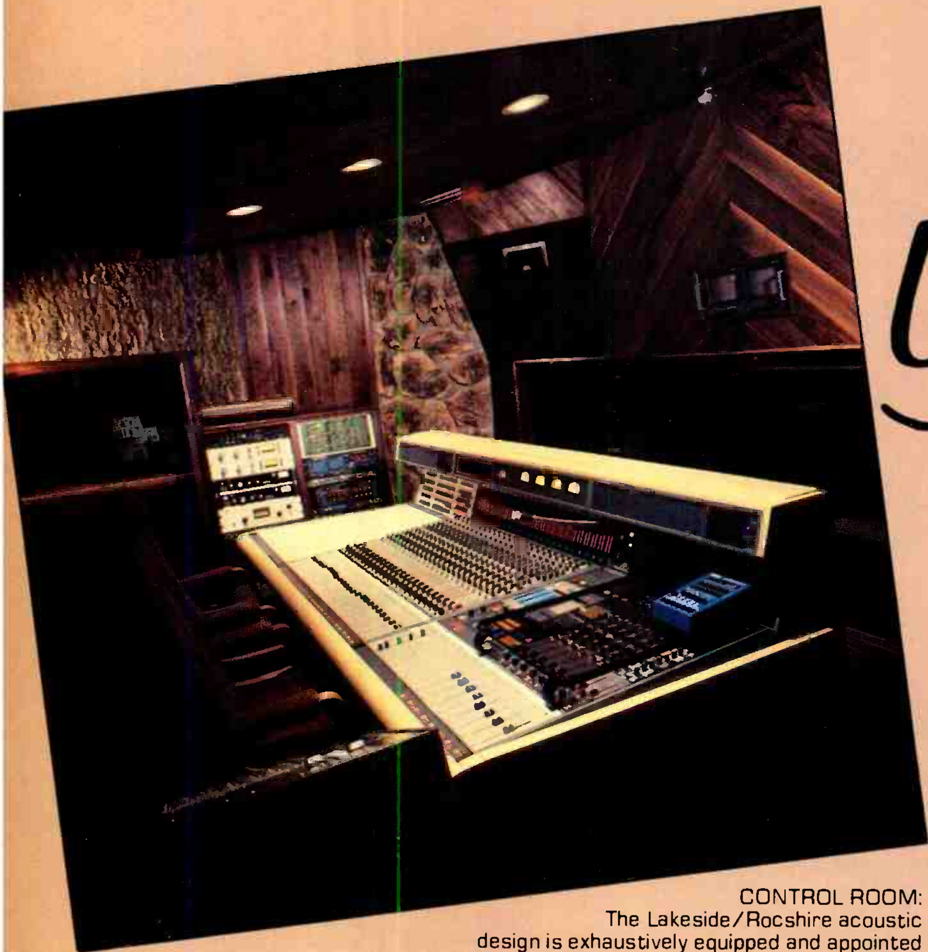


Figure 1: First- and Second-Level CADP Menus.

Producers . . . Engineers . . . Performers . . .



CONTROL ROOM:
The Lakeside/Rocshire acoustic design is exhaustively equipped and appointed around a NEVE 8128 32x24 (capable of 64 lines in) control desk; a unique, instantly interchangeable from the console, system of monitors giving the engineer/producer the choice of TAD/JBL, UREI 813 Time Align™, Tannoy SR12 monitors; the most complete outboard complement including a Publison with keyboard, and Lexicon 224X.

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SYNTHESIZERS & KEYBOARDS:
Studio Manager, Lester Claypool with dual Roland Jupiter 6s, E-mu Emulator, Oberheim OBX-A, OBX-8, DMX Drum Simulator, and DSX Drum Sequencer, Fender/Rhodes among other keyboards, all of which are accompanied by Rocshire's two on-staff programmers.

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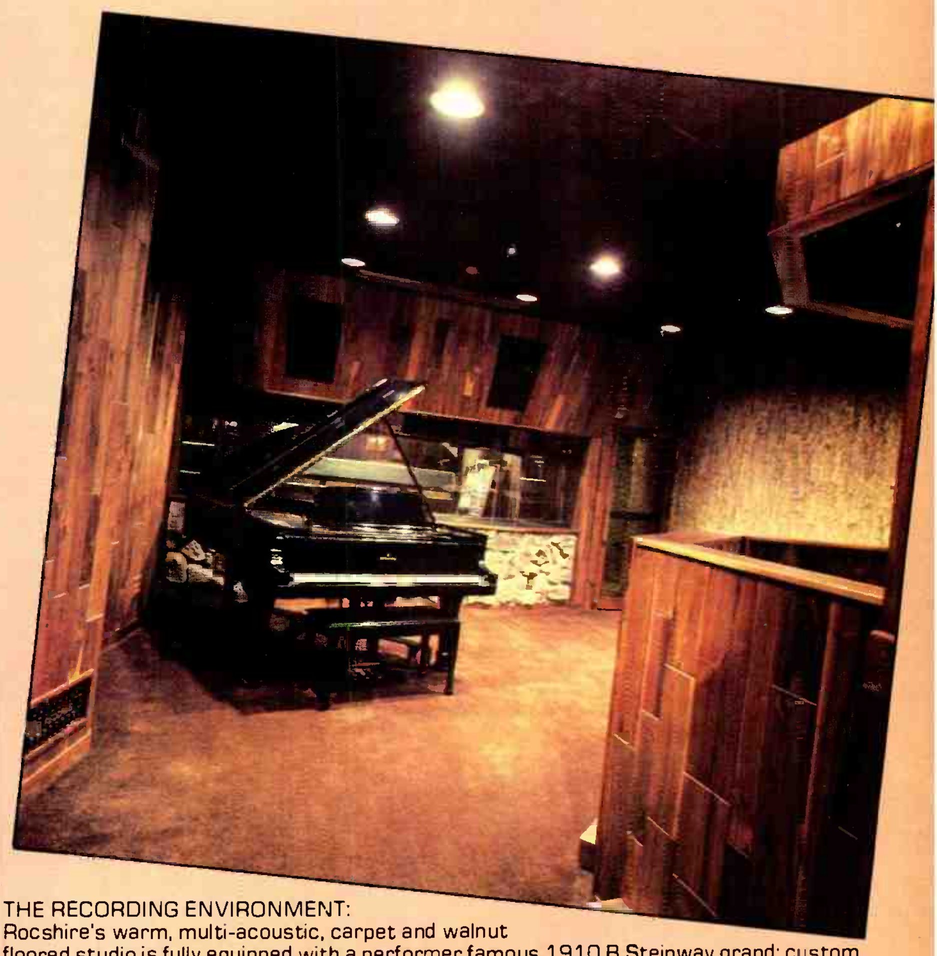
Exemplified by Tony Carey's premier album . . . carefully nurtured by Rocshire's in-house marketing and promotion staff, his first single rose to #1 AOR airplay (Radio & Records). The second single is currently at 70 with a bullet on the Billboard chart.



THE RECORDING ENVIRONMENT:
Rocshire's warm, multi-acoustic, carpet and walnut floored studio is fully equipped with a performer famous 1910 B Steinway grand; custom made studio-compatible Charvel/Jackson Guitars and Bass; full Simmons synthesizer drum kit, in addition to Ludwig and Tama kits. The microphone complement includes a large variety of classic tube types, including the Telefunken 251, Neumann U47s, M155s, M269, KM86, U67s, and stereo SM69s.

For additional information circle #26

For additional information circle #28

CADP PROGRAM

. . . continued from page 40

Module is a rectangular approximation of the space, and the user enters its three dimensions, along with absorption coefficients for the six sides. The program then calculates the acoustical data for the specified space. More than one room module can be used in approximating more complex spaces.

Working from a floor plan, the user is asked to enter the Cartesian coordinates for a seating plane or planes, thus allowing complex surfaces to be entered.

Loudspeaker Menu

Having entered the location of the seating plane or planes, together with acoustical information for the space being studied, the loudspeaker parameters now can be specified. In this menu, the following are requested:

1. Loudspeaker type.
2. Frequency band.
3. Mounting location (x, y, and z coordinates).
4. Rotation angle.
5. Azimuth angle.
6. Elevation angle.
7. Relative drive level.

Up to 20 loudspeakers may be specified, and independently aimed and powered. Each loudspeaker in the array is given a working name, such as S1, S2, etc., so that individual loudspeakers can be deleted or re-aimed as desired.

Display Menu

At this point, all pertinent data on the array and its location with respect to the audience in a given acoustical environment has been entered into the program. We are now ready to look at the system's performance. Figure 2 shows what happens when the program examines the normalized direct-field response on the seating plane. Displayed on the IBM video monitor will be the seating area in plan view, scaled by the program to make the best fit into the available screen space. The program calculates the inverse square losses at points on the seating plane, as seen through the horn's polar pattern. Such parameters are computed for as many "slots" as are available on the screen; typically, there will upwards of 100 points read out on the monitor.

The numbers displayed on the monitor have been normalized by the program — that is, the highest value will be zero dB, with all other values negative with respect to it (see accompanying sidebar).

Merging of multiple loudspeaker directional patterns is carried in two ways, which will be discussed in detail later.

The direct-to-reverberant ratio at the seating area can be examined, since it is simply the difference in the direct field levels and the reverberant field level as calculated by the following equation:

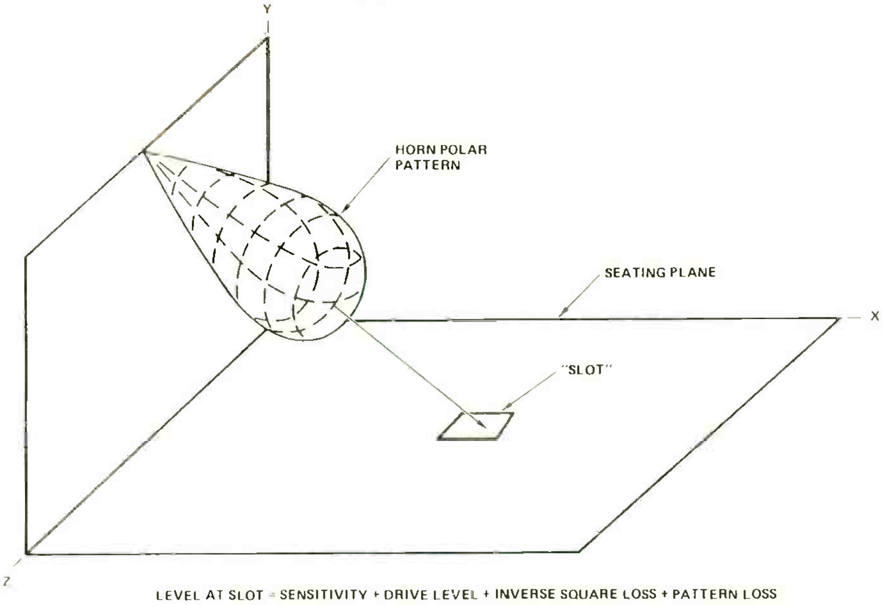


Figure 2: Horn Aiming and Direct-Field Calculation.

$$\text{Reverberant Level (dB)} = 126 + 10 \log (W/R)$$

In this equation, W, the total acoustical power output of the array, is calculated from the drive levels of the components, and the efficiencies of the transducers. R, the room constant, is given by the following equation:

$$R = S\bar{a}/(1 - \bar{a})$$

In sound reinforcement systems, most of the sound from the loudspeakers is first incident on the audience, with its relatively high absorption coefficient. This tends to diminish the level of the reverberant field, and generally improves intelligibility. In the CADP program, the assumption is made that roughly two-thirds of the sound from the array is incident on the audience, while one-third strikes the walls. Thus, a variant of the room constant is used, called R', defined by the following equation:

$$R' = S\bar{a}/(0.4 - \bar{a}/3)$$

This equation is a modification of the equation for R' as developed by George Augspurger (see: *JBL Sound System Design Reference Manual*; pages 5-18 to 5-21), and assumes that the absorption coefficient for the occupied audience area is 0.9 in the 1 to 2 kHz range.

A choice is offered of viewing a display of the direct-to-reverberant ratio using R' or R. The display using R' is a more accurate picture of the direct-to-reverberant ratio at the seating plane when the seating area is fully occupied.

We can then move on to consider the display of estimated intelligibility. Here, the direct-to-reverberant ratio calculated by using either R or R' is compared, point by point, with the reverberation time. The graphs shown in Figure 3 are derived from Peutz' data on intelligibility, as shown in the *JBL Sound System Design Reference Manual*, pages 6-14 and 6-15. The display shows the estimated intelligibility in four zones: *EX* (Excellent); *GD* (Good); *OK* (Acceptable); and *QU* (Questionable). These zones correspond to intelligibility estimates as follows:

Probable Articulation Loss of Consonants
Excellent: Less than 5%
Good: 5% to 10%
Acceptable: 10% to 15%
Questionable: Greater than 15%.

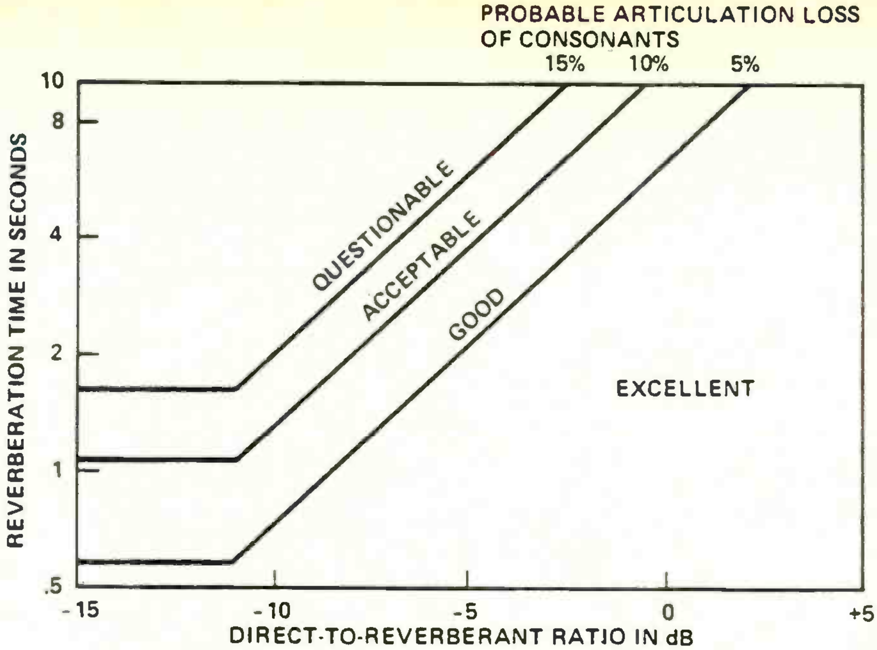


Figure 3: Calculation of Estimated Intelligibility.

The maximum direct soundfield also can be examined. This parameter is simply the direct field levels read on the seating plane when the array is powered to its normal limit.

The mechanical design display provides front, side, and top views of the array, and a small red dot on the monitor indicates the center of gravity of the array as an aid in designing rigging. Other options in this part of the program are variable scaling of the views, and views of the individual elements in the array.

The Hard Copy Menu prints out all room dimensional and acoustical data, as well as all mounting data on each element in the array.

At several points in the program's menu structure there are Retrieve Data

— continued on page 49 . . .

SAMPLE PRINTOUTS PRODUCED BY THE CENTRAL ARRAY DESIGN PROGRAM

Shown below are several hard-copy program printouts to illustrate the flexibility of CADP. The space analyzed was a room with a 16- by 40-meter seating area, and height of 20 meters (approximately 52 by 130 by 65 feet). The loudspeaker array is located in the center of the left wall at a height of 20 meters. The hard-copy printouts detail the room's dimensional and acoustical characteristics, along with details of loudspeaker type, location, orientation, and powering.

In Figures A through E, the merging program that assumes coherent radiation from all the elements in the array was used. Figure A shows the Normalized Direct Field; Figure B the Maximum Direct Field; Figure C the Direct-to-Reverberant Ratio for R'; and Figure D and

Normalized direct field. Press Enter--

-3	-3	-2	-1	-2	-2	-3	-4	-5
-5	-4	-3	-1	-1	-3	-4	-4	-5
-4	-2	-2	-1	-1	-2	-3	-4	-6
-4	-2	-1	-1	-2	-2	-3	-4	-5
-5	-3	-1	-1	-2	-2	-3	-5	-5
-4	-3	-2	-1	-2	-2	-3	-5	-5
›5	-2	-1	-1	-1	-2	-3	-5	-5
-5	-2	-1	0	-1	-1	-3	-4	-5
-3	-3	-1	-1	-2	-2	-3	-5	-5
-3	-3	-2	-2	-2	-3	-2	-5	-5
-3	-1	-1	-1	-2	-3	-3	-4	-5
-3	-1	-1	-2	-1	-3	-3	-4	-5
-4	-2	-2	-2	-1	-2	-3	-4	-5
-3	-2	-3	-2	-2	-2	-4	-4	-5
-3	-3	-2	-1	-2	-2	-4	-4	-5

5 m

2kHz

Figure A: Normalized Direct Field

Maximum direct field. Press Enter--

107	107	108	108	108	108	106	106	105
105	106	107	109	109	107	106	106	105
106	107	108	109	109	107	107	106	104
106	108	109	109	108	107	107	106	105
105	107	109	109	108	107	107	105	104
106	107	108	109	108	108	107	105	104
105	108	109	109	109	108	106	105	104
105	108	109	109	109	109	106	106	104
107	107	109	109	108	108	107	105	104
107	107	108	107	108	107	107	105	104
107	109	109	108	108	107	107	106	105
107	109	109	108	109	107	107	106	105
106	108	108	108	109	107	107	106	105
107	107	107	108	108	108	106	106	105
107	107	108	108	108	108	106	106	105

5 m

2kHz

Figure B: Maximum Direct Field.

Direct/reverb. for R.' Press Enter--

-4	-4	-3	-3	-3	-4	-5	-5	-6
-6	-5	-4	-3	-2	-4	-5	-5	-6
-6	-4	-4	-2	-2	-4	-4	-5	-7
-5	-3	-2	-2	-3	-4	-4	-5	-6
-6	-5	-2	-2	-3	-4	-4	-6	-7
-5	-5	-3	-2	-3	-3	-4	-6	-7
›7	-4	-2	-2	-2	-3	-5	-6	-7
-6	-4	-2	-2	-2	-2	-5	-5	-7
-4	-4	-2	-2	-3	-3	-4	-6	-7
-4	-5	-3	-4	-4	-4	-4	-6	-7
-5	-2	-2	-3	-4	-4	-4	-5	-6
-4	-3	-2	-3	-2	-4	-4	-5	-7
-5	-3	-4	-3	-2	-4	-4	-5	-6
-4	-4	-4	-3	-3	-3	-5	-6	-6
-4	-4	-3	-3	-3	-3	-5	-6	-6

5 m

2kHz

Figure C: Direct-to-Reverberant Field For R′.

Est. intelligibility (R).Press Enter--

OK	OK	GD	GD	GD	GD	OK	OK	QU
QU	OK	OK	GD	GD	GD	OK	OK	QU
OK	GD	GD	GD	GD	GD	OK	OK	QU
OK	GD	GD	GD	GD	GD	OK	OK	QU
OK	OK	GD	GD	GD	GD	GD	QU	QU
OK	OK	GD	GD	GD	GD	GD	QU	QU
QU	GD	GD	GD	GD	GD	OK	OK	QU
QU	GD	GD	GD	GD	GD	OK	OK	QU
OK	OK	GD	GD	GD	GD	GD	OK	QU
OK	OK	GD	GD	GD	GD	GD	OK	QU
OK	GD	GD	GD	GD	GD	OK	OK	QU
OK	GD	GD	GD	GD	GD	OK	OK	QU
OK	GD	GD	GD	GD	GD	OK	OK	QU
GD	GD	GD	GD	GD	GD	OK	OK	QU
GD	GD	GD	GD	GD	GD	OK	OK	QU

5 m

2kHz

Figure D: Estimated Intelligibility for R′ (Room nearly empty).

CENTRAL ARRAY DESIGN PROGRAM SAMPLE PRINTOUTS ... continued —

E the Estimated Intelligibility (ranging from *Excellent* to *Questionable*) for both an empty room (Figure D, using R), and a full room (Figure E, using R').

Figures F thru H show the computed layout of the cluster top view, front view, and side view, respectively.

As an example of the more complex merging program, consider Figures I and J. In Figure I, two JBL 2380 horns are placed side by side, their drivers separated by a horizontal distance of 0.5 meter (20 inches); the test frequency is 2 kHz. Note the appearance of a major lobe along the axis of the two loudspeakers. The null angles are clearly defined (note dashed lines), and minor lobes can be seen outside the dashed lines.

Figure J shows the same two loudspeakers, this time separated vertically by 0.5 meter. Note the appearance on the seating plane of null and reinforcement zones. A dashed line shows a reinforcement zone, and a dotted line denotes a cancellation zone. ■■■

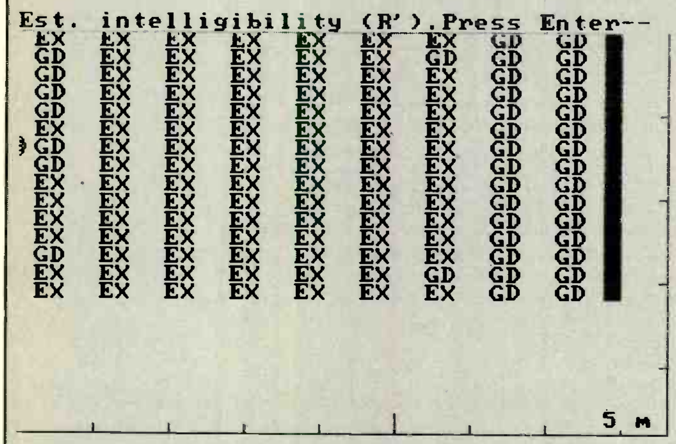


Figure E: Estimated Intelligibility for R' (Room fully occupied).

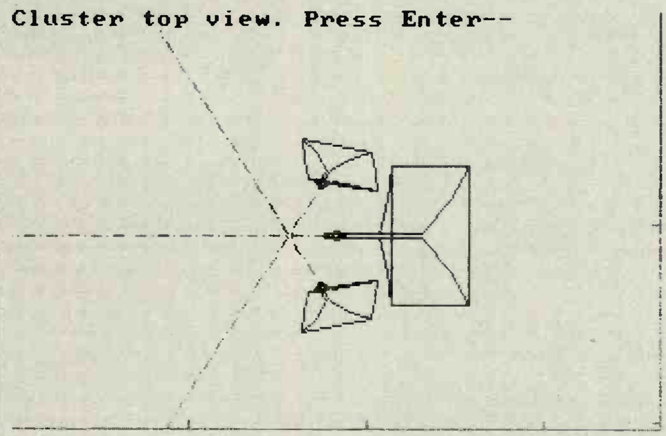


Figure F: Specified Loudspeaker Cluster viewed from above.

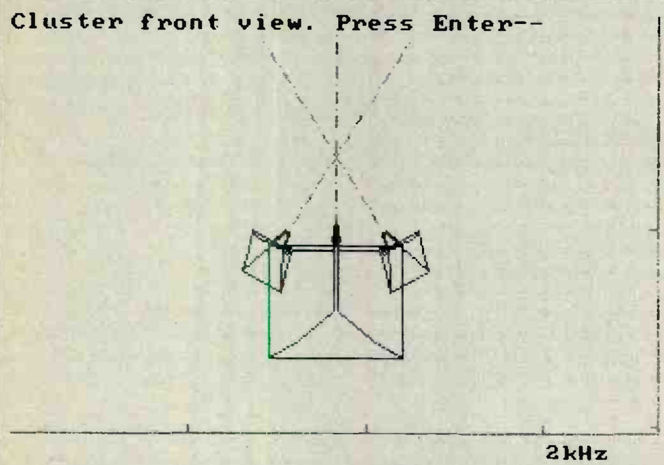


Figure G: Specified Loudspeaker Cluster viewed from front.

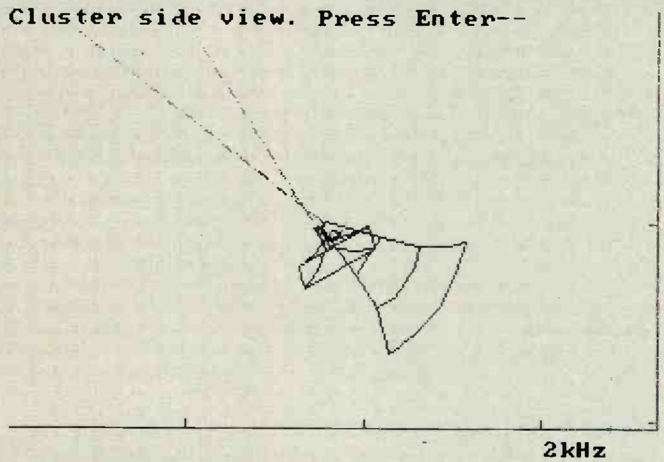


Figure H: Specified Loudspeaker Cluster viewed from side.

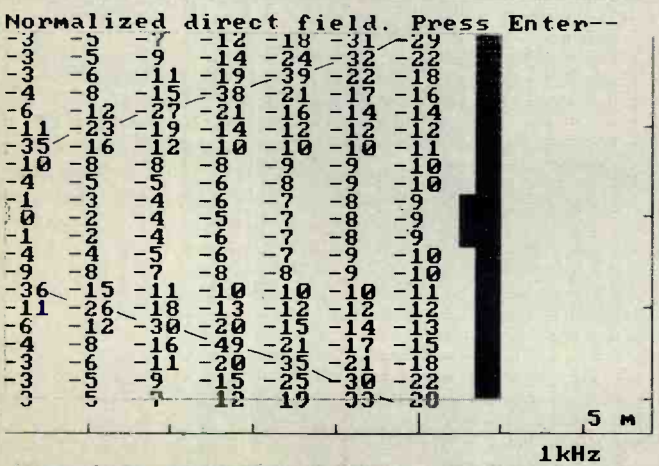


Figure I: Normalized Direct Field for two JBL 2380 horns separated horizontally by 20 inches. (Dashed lines show null angles.)

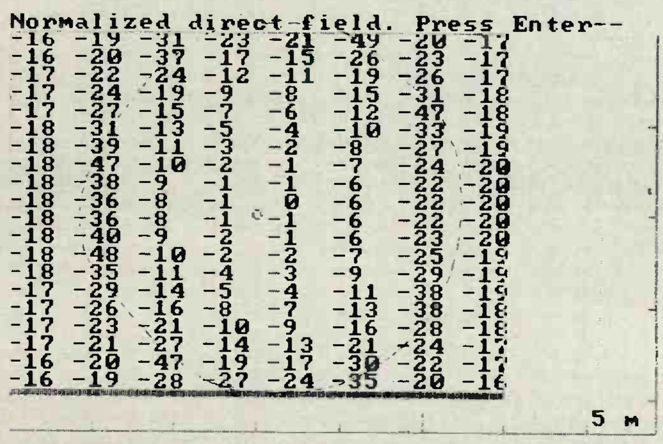


Figure J: Normalized Direct Field for two JBL 2380 homs separated vertically by 20 inches. (Dashed lines show reinforcement zone; dotted a cancellation zone.)

. . . continued from page 46 —

and Save Data options, which enable the user to assign a name to a given job, and store or recall partial data at any point in the program. The Main Menu contains the option for ending the job session, and committing all dimensional and acoustical data to the data diskette.

Other Features

A "Print-Screen" program allows the data on the monitor screen to be printed out at any time. When this is requested, the dot matrix printer prints each picture element (pixel) being displayed on the video monitor. This feature is especially useful in the mechanical design part of the program, since printouts can be given directly to a design draftsman for final drawing of the array.

Merging of Directional Patterns

Since up to 20 elements can be put into the array, there must be consistent rules governing their interaction. In CADP, the designers have opted for two pattern-merging strategies. The first of these assumes that all elements radiate coherently; it is a reasonable assumption if all the acoustical centers of the devices are located on a spherical surface, and the drivers lie parallel to that surface.

The second option is more complex, and takes into account phase as well as amplitude relationships at various frequencies, thus allowing line arrays to be modeled and displayed. Both sets of computations are made simultaneously, and the program prompts the user to select one option or the other.

The Horn-Pattern Merging programs take into account the relative drive levels of the horns. In essence, the merging programs generate a Horn Directional File of frequency-dependent parameters for each horn in the cluster, and then consider the rotation, azimuth and elevation, and drive level contributing towards the measurement "slot" under consideration. These contributions are then summed to provide the resultant overall pattern. Also, up to 20 single horns can be set up in a linear array and then combined via the phase merging program to model a line of units arranged, for example, across the front of an auditorium, rather than in a central cluster.

* * *

As will readily be appreciated, the Central Array Design Program developed by JBL for the IBM personal Computer is intended to reduce the tedious and often time-consuming computations necessary to calculate the coverage pattern and intelligibility of multiple-array sound reinforcement systems. While the CADP cannot design the system for you, careful application of computer-assisted procedures such as these can be of great help to the sound-system design engineer and acoustics consultant. ■■■

All photography by KATHY COTTER

WESTLAKE STUDIO C
APPLICATIONS OF CONTROLLED TRAVEL PATH DESIGN

By Mel Lambert and Paul Laurence

When the time came in 1981 for Westlake Audio to move the original "Mix Room," located formerly at the company's Wilshire Boulevard premises in Los Angeles, it was considered that the construction of just a remix facility would not be cost-effective in today's competitive studio market. Instead, the decision was made to build a state-of-the-art control room and studio combination — christened Studio C, to follow in logical sequence from Studios A and B at its West Hollywood facility — in Westlake's new building on Santa Monica Boulevard. Ground breaking for the new studio, office, and demonstration complex took place in the summer of 1982, and by Christmas of last year the new studio was open for business.

So far the new facility has played host to several illustrious clients, including Earth, Wind & Fire, EWF's producer Maurice White working with Jennifer Holliday (star of the broadway play, "Dreamgirls"), Doc Severinson, and several Japanese clients.

Compared to previous Westlake installations — including such noteworthy examples as Mr. "C," Omaha; George Duke's and Georgio Moroder's personal-use facilities in Los Angeles; Electric Lady in New York; and Crown Records, Hong Kong — design of the recently completed Studio C, company president Glenn Phoenix would be the first to acknowledge, represents a less lavish approach. "Studio C is not as fancy in terms of finish materials," he concedes, "but rather concentrated on the 'high-tech/high-resolution' aspects of recording sound, while still maintaining a high level of finish and professionalism to the environment."

Constructed within one end of a 15,000-square-foot building that formerly served as a lighting warehouse, the new facility comprises a 900-square-foot recording area, linked to a 500-square-foot control room; space also is available elsewhere within the complex for a second studio, to be known as Studio D, which currently is still in the initial planning stages.

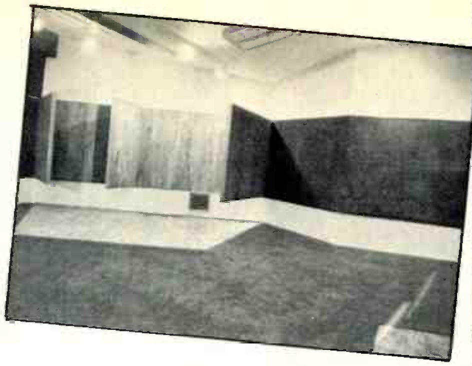
Control Room Design and Monitoring System

The control room itself is large and spacious (ceiling height varies between 7 and 13 feet), and can comfortably accommodate up to 10 people — a maximum of five behind the vintage API console; three in the playing/listening area in front of the board; and a couple of musicians playing direct elsewhere in the room. If, on initial impressions at least, the control-room layout and acoustics treatment appears somewhat unconventional — the walls and ceiling angles are rather steep, and the presence of an eight- by four-foot skylight in the ceiling takes some getting used to — the unfamiliar look and feel of the environment can be attributed to Westlake's Controlled Travel Path Design.

While an accompanying sidebar to this article provides full details of the CTP design philosphy, in essence it attempts to reduce the number of early sound reflections arriving at the central monitoring area. CTP pays particular attention to the angles of reflection and refraction of sound coming from the monitor loudspeakers and secondary sources, in an attempt to minimize the number of repetitive travel paths around the room. This precaution, coupled with

close attention to the selections of surface acoustic treatments, and maintaining rigid boundary walls to the environment, Phoenix says, produces a broad, high-definition soundfield across the width of the console monitoring position.

Given the control room's large volume, a loudspeaker system had to be custom designed by Westlake to handle the high power necessary for adequate monitoring levels. Developed specifically for Studio C, the new Westlake SM-1 is a five-way/quad-amplified system housed in a massive 38 cubic feet enclosure (making it possibly one of the largest control room monitors currently in use), and is capable of delivering an impressive SPL into the room. According to Westlake chief technician and general manager, Jim Fitzpatrick, a pair of JBL Model 2245H 18-inch bass drivers handle the bottom end (and are capable, Westlake claims, of providing approximately 6 dB more low-frequency headroom that dual 15-inch units), while a single TAD Model TM-1201 12-inch driver covers the mid-bass region. (RMS power handling of the low-frequency woofer system is a total of 1,200 watts across the four 18-inch drivers, Fitzpatrick adds.) Mids and highs are covered by a JBL Model 2441 two-inch midrange compression driver on a proprietary solid walnut horn assembly; a JBL Model 2421 one-inch upper-midrange compression driver on a second proprietary horn; and a JBL Model 076 half-inch throat compression "Super-Tweeter," passively crossed over for the high frequencies.



Above: hinged, double-sided absorbent/reflective panels line one side of the studio; **Right:** view from studio to control room, with bare brick wall to right; **Below:** large, electrically operated skylight.

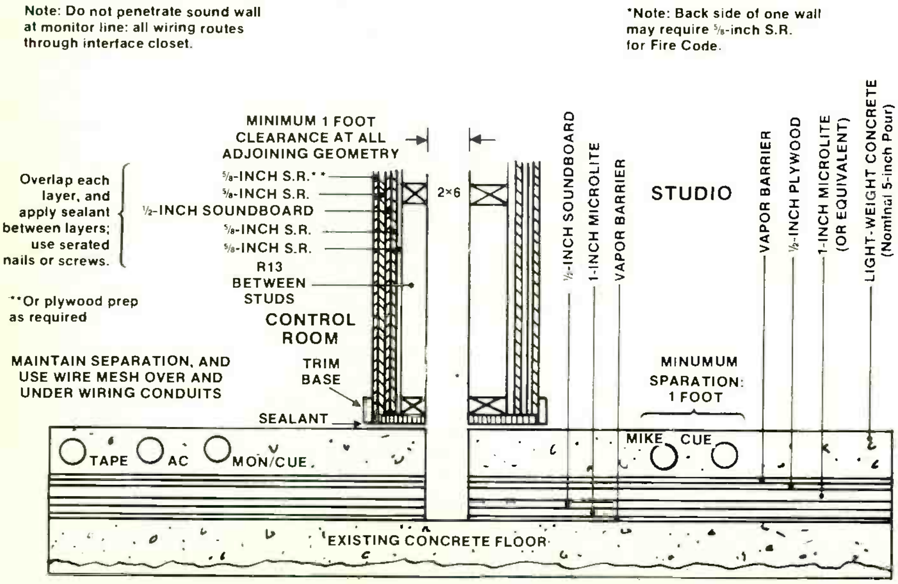
Westlake Audio — Company Origins

Current company president Glenn Phoenix, who took on the position when Tom Hidley moved to Europe in 1975, describes Westlake as a full service sales and design company specializing in studio contracting and design, and which "emphasizes equally the sound of a room and its physical environment." In practical terms, the company puts special emphasis on three main areas: high-resolution monitor loudspeakers that are multi-amplified for low distortion, and set up in a multiway drive configuration for controlled dispersion, low distortion, and phase coherence; good design principles, such as the innovative Controlled Travel Path or CTP™ approach; and good implementation via experienced construction crews, and the selection of appropriate materials.

Many consider the "Westlake Look and Sound" to be the characteristic studio design of the Seventies and early Eighties — active trapping utilizing relatively small-size rooms that, in terms of acoustics, offered controlled dispersion, even RT60, and good sound isolation of extremely high SPLs. These characteristics also were favored for their consistency of sound; the original Record Plant studios in New York, Los Angeles, and San Francisco were designed to have an identical sound characteristic, so that tapes recorded at one studio would sound virtually identical in any other Westlake/Eastlake room.

Westlake probably is best known, however, for having removed the "institutional feel" from studios; beyond being just physically attractive and comfortable places in which to work for prolonged periods of time on a lengthy multitrack project, these studios were intended to reach out to all the senses. Most of all, the company attempted to fabricate studios in which an artist and producer could live, create, play, and make high-quality recordings. ■■■

DETAIL OF ACOUSTIC TREATMENT AT CONTROL ROOM/STUDIO BOUNDARY



Why did Westlake opt for a multiway speaker system, when the majority of studio designers and monitor manufacturers now are making much of the sound performance capabilities of two- and three-way systems, we queried? "It's an inevitable fact of physics that you'll end up with noticeable distortion and dispersion problems if you attempt to drive any speaker wideband," Phoenix offers, "despite what some of the manufacturers would have you believe.

"But I'll concede that other multiway systems have gotten a bad name in the industry for being expensive, and not sounding very good. But sound problems are not inherent in multiway monitors — they are rather the result of poor design.

"There are two areas, in particular, where other designs have fallen down: the speaker's mechanical configuration, which often prohibits a symmetrical horizontal polar pattern; and crossover units, which typically produce non-coincident waveforms throughout the crossover areas. To our knowledge, the SM-1 is the highest power, phase-coherent monitoring system currently available to studios."

To ensure correct multiway operation, the SM-1 system is configured to operate with a custom-designed Westlake crossover unit, which has been tuned specifically to suit the room's individual speaker components and physical layout; crossover frequencies are set at 200 Hz, 800 Hz, and 3.2 kHz (passive HF crossover occurring at 10 kHz). Housed in a 19-inch rackmount case, the high-resolution crossover unit contains two channels of four-way, 24 dB per octave filters, which minimize driver-to-driver interferences, and optimize bandwidth control. (A matching 24 dB per octave high-level crossover unit in each speaker cabinet handles the passive high-frequency transition.)

As Glenn Phoenix explains, "The 24 dB per octave slopes are used for the crossover because you need to get the selected signal band into — and out of —a speaker driver quickly, while maintaining complementary phase, both above, at, and below the crossover points. This allows for a virtually seamless transition of sound from one driver to the next."

In addition, appropriate delay compensation is built into each crossover drive amplifier to suit the geometry and layout of the particular system, and ensure correct phase operation of the five drive units.

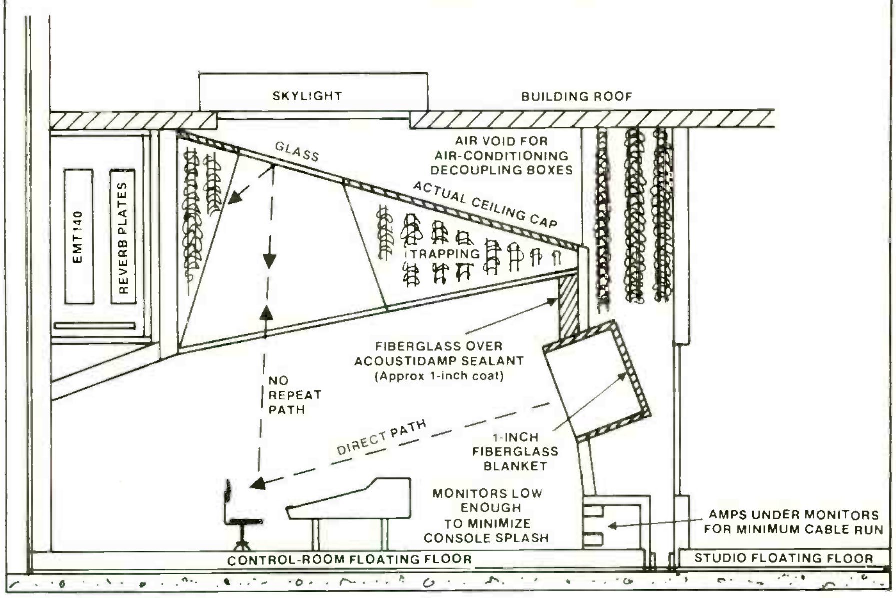
Also provided in the control room for center-channel film dubbing and mixing applications is a single Westlake TM-6 two-way speaker, connected to the monitoring system via a custom split box. "Close-field" monitoring requirements can be covered with a choice of Westlake BBSM-6, Yamaha NS-10, Tannoy SRM-10B and -12B, JBL 4311, 4312, and 4411 units for close mounting on top of the console meter bridge.

And the creative atmosphere hasn't been neglected either. The control room offers dimmable incandescent lighting, florescent work lights, plus a novel electrically controlled skylight that enables natural sunlight — or moonlight, for that matter — to enter the room. The skylight, a feature of the original building that Westlake decided to incorporate into the design of Studio C, is sealed off from the control room by a pane of thick glass, which has been positioned carefully to function as part of the acoustic design.

Recording and Processing Equipment

At the heart of the control-room recording hardware is the original API 32-input console from the Mix Room, dating back to circa 1974, and which has undergone extensive modification and upgrading over the years. Since an increasing proportion of the studio's sessions now involve 46-track remix work, the API has been modified so that the 24-input monitor section can be patched and routed directly to the stereo and quad busses, in addition to the main 32-input section. The board now offers four independent echo and delay send/return busses, with individual level controls from each input module switchable to the required delay busses. A passive output level control has been added to each of the multitrack busses so that they can be used as additional sends during complex remix dates; the multitrack sends also can be selected pre- or post-channel fader, which removes the channel VCA from the circuit during tracking.

CROSS SECTION OF CONTROL ROOM WITH SKYLIGHT AND CEILING DETAIL



Other modifications Westlake has made to the API board include the fitting of Jensen nickel-core transformers to certain EQ sections and output busses, along with Hardy 990 op-amps to a number of the multitrack, stereo, and monitor outputs — the idea being, Fitzpatrick explains, to offer a variety of sound-quality options to the engineer or producer, in terms of being able to select preferred devices and outputs on the board.

Studio C's pair of 3M M79 Series 24-tracks also have been extensively refurbished. One of the pair is permanently assigned to the room, and has had all its input and output transformers removed, ICs upgraded for enhanced slew rate, and the transport logic modified for improved interlock with the facility's BTX Shadow SMPTE synchronization system. Since the second M79 multitrack also serves as a "float" machine for the rest of Westlake's studios, and is available for rent to outside clients, it still retains its input and output transformers; as Fitzpatrick offers, you often run into potential ground-loop problems in a unknown control-room environment if a machine isn't adequately isolated via transformer circuitry.

Mastering duties are handled by a selection of Ampex ATR-102/4 quarter- and half-inch, two- and four-track decks, plus Studer A80-VU and 3M M79 transports. (Also available at extra charge is a 3M 32- and 4-track Digital Mastering System, and companion Editing System.)

To handle the increased demands on outboards and effects during 46-track sessions — not to mention covering today's complex 24-track mixdowns — the studio boasts an impressive collection of sound benders and blenders. DDLs and effects include a Lexicon Model 93 Prime Time and Model 97 Super Prime Time, a Publison Fullmost

PHILOSOPHY OF WESTLAKE'S CONTROLLED TRAVEL PATH DESIGN

Westlake Audio president Glenn Phoenix would be the first to acknowledge that his innovative design for the new Studio C at the facility's Santa Monica Boulevard, Los Angeles, location is based upon a distillation of the main acoustics theories and practical construction techniques developed over the last decade or so. But with some interesting and revolutionary advances and fundamental improvements.

While the recording area purposely features a reasonably live acoustic treatment — a trend that is appearing in an increasing number of studios these days — it is in the control room design and implementation that the fundamental differences between the new Westlake and "traditional" designs lie.

"The philosophy behind our new 'Controlled Travel Path,' or CTP design," Phoenix says, "is that the angles of reflection and surface acoustics treatment are arranged to restrict early sound reflections into the listening or monitoring environment. The result is a broad, high-definition soundfield across the central area of the control room, with a RT60 that is consistent at about 0.3 seconds within the low- and mid-frequency range, falling to around 0.25 seconds above 8 kHz."

The underlying key to the new design approach, Phoenix offers, is to minimize the repetitive paths for sound reflection around the surfaces of the control room, including both standing waves between parallel walls, floor, ceiling, and other areas, plus the odd reflection and refraction modes between room corners, etc. By controlling the ways in which sound waves from the monitor loudspeakers and secondary sources can repeatedly "bounce" or be reflected and refracted around the room, a uniform, well-behaved environment possessing a diffuse soundfield with a consistent decay characteristic can be created.

And if there appears at first sight to be a close kinship with at least the fundamental approach of Chips Davis and Syn-Aud-Con's Live-End/Dead-End™ design philosophy — in which, in essense, the console monitoring position defines a line across the control room in front of which the acoustic treatment is purposely highly absorbent, while behind it is configured to be reflective — Phoenix would readily agree. But with certain important reservations, however.

"To me," he concedes, "the LEDE design creates a 'psychological uneasiness' for a producer or engineer working in the room. It often feels unnatural to monitor in an environment where there are absolutely no sound reflections coming from the area between you and

Westlake SM-1 five-way monitors, and sound-absorbent wall treatment.

processor (currently being used, Fitzpatrick relates, by producer Maurice White to create some interesting vocal effects), an Eventide H949 Harmonizer, plus Sontec and GML parametric equalizers. Available echo/reverb units include a pair of stereo EMT 140 plates (one with tube electronics, and the other solid-state), plus EMT 250, Lexicon 224X, and Eventide SP2016 digital reverb devices. Other signal processors include full Audio + Design Recording Scamp and dbx Series 900 racks, Inovonics Model 201, Sontec, dbx, GML, plus UREI LN-1176, LA-3, and LA-4 compressor-limiters.

Recording Studio Acoustics

Designed to accommodate between 10 and 25 musicians on its 900 square feet of usable floor space, the studio proper has an average ceiling height of 12 feet, making it a suitable venue for string and horn dates, as well as traditional sessions.

"Studio C has a more 'live' acoustics feel than most of Westlake's previous room," Phoenix says. "It was designed to be reasonably neutral in sound characteristic, yet allow medium to long decay times — between half and one second — to be set up, should an engi-

"Close-field" monitors atop the console include Tannoy and Auratone units.

WESTLAKE CTP DESIGN PHILOSOPHY . . . continued —

the loudspeakers, while the space behind you is highly reflective.

"The Controlled Travel Path design recognizes the importance that early reflections play in the way we perceive a complex soundfield — as does LEDE — but the practical result is an environment in which as you move around the room there is an even and smooth transition from live to absorbent surfaces, rather than a steep and narrow one.

"Our approach is to analyze the directions that a wavefront radiating from the monitors can take as it travels into the room, and attempt to configure the acoustic design so that the reflections fall within an area of the room that causes the minimum number of interference paths due to those early reflections."

As well as placing a greater emphasis upon the behavior of sound reflections around the room, Phoenix points out that Studio C is somewhat different from previous Westlake and Eastlake/Sierra environments. (The former rooms having been designed by both Westlake founder Tom Hidley and, since 1975, Glenn Phoenix's team, following Hidley's move to Europe, and the resultant company name change.) The main design departure is the apparent acoustic nature of the studio and control room, which would be described by most users, he suggests, as a live-sounding studio area, coupled to a reasonably dead-sounding control room. While the measured RT60 of Studio C is indeed higher than most recording studios — a 0.5 to 1 second decay can be achieved, depending on the room setup and frequency of interest — the control-room reverberation time is practically identical to that encountered in conventional spaces. The apparent perceived difference in the sound characteristics of the CTP design, he offers, can be attributed to the lack of early sound reflecctions, and controlled angles in the room, which produce a diffuse soundfield, particularly at low frequencies.

While calculating the physical geometry of the control room necessary to minimize early sound reflections into the monitoring environment, Phoenix relates that care needs to be taken in the selection of materials that eventually will be used as surface treatment, and the frequency of sound under consideration.

"A portion of the sounds incident on a surface will be absorbed by the structure; some will be reflected at an equal but opposite angle — as with light on a mirror; some will be redirected due to diffraction — depending on the surface and wavelength being considered; while some may be diaphragmatically reradiated into the listening area — usually the low frequencies, and possibly parallel to the surface, regardless of the impinging angle. This parallel radiation

— *continued* . . .

WESTLAKE CTP DESIGN PHILOSOPHY . . . continued —

or diaphragmatic phenomenon is caused by the fact that surfaces never possess 100% rigidity; if they did, we would be able to presume that all the sound will follow the simple 'angle of reflection equals the angle of incident' rule, and predict the way early reflections behave in the room by simple geometric observations. But, since the problem is more complicated than that — particularly at low frequencies, because of the long wavelengths involved — a designer needs to be aware of the diaphragmatic action of the surfaces involved.

"You can do a lot of things to minimize such diaphragmatic action, by adding mass to the walls, or controlling the way that the mass and wall system work together, either as a homogeneous unit or not. And these diaphragmatic effects can produce noticeably different sound characteristics in identical or similar rooms. They can be caused by surface treatment at mid- to high-frequencies, or by the whole wall system acting as a large diaphragm at long wavelengths."

As a result, Phoenix offers, when designing larger rooms, such as the control room for Studio C, the walls tend to be more diaphragmatic at lower frequencies, since the larger areas involved can flex more easily. To help prevent such physical movement, or at least minimize it as far as possible, another major factor in the CTP design is to strive for a condition of minimum resonance; in other words, ensure that the outer walls of the control-room structure are braced and reinforced, so that the result is a rigid set of external boundaries with which the sound will interface. (And which, of course, also will ensure adequate isolation for sound both trying to leave and enter the monitoring environment.) Boundary rigidity is achieved through the use of integral triangulated bracing, and homogeneous, high-mass wall and control-room glass systems.

It goes without saying that in order to produce a diffuse and even soundfield across the monitoring area, the primary consideration is one of irregular geometry — so that parallel surfaces are eliminated — but with equal emphasis, Phoenix points out, on studying very carefully the physical boundaries and layout of the room shell, and the sound travel paths. By way of an example, he says, "consider the odd angles in the overhead skylight. Although, to the casual eye, the external geometry may appear to be sufficiently irregular to prevent repetitive sound reflection paths, we need to take note of the actual angle 'seen' by the sound itself as it is reflected and refracted off each surface on its passage around the room."

An often overlooked travel path, Phoenix says, is that of structure conduction. To minimize this phenomenon, which can cause a considerable amount of distorted and out-of-phase sound to be reradiated into the listening area, several isolation breaks often are employed to reduce the amount of energy conducted to potential radiators. These breaks usually take the form of speaker isolation mounts, and wall and floor breaks.

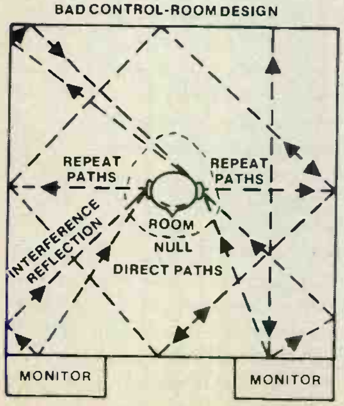
"Once we have worked out the dimensions and angles needed to secure a diffuse soundfield across as wide a monitoring area as possible, we then look at the absorption characteristics of the materials in and around the room. The normal materials — fiberglass and sound blankets, plus membrane resonators and sufficient bass trapping — were selected, but more important was where in the room they need to be positioned.

"Calculating and designing a CTP room is a slightly more lengthy process than a 'conventional' design," Phoenix concludes, "and cannot be accomplished successfully in some spaces. It takes more experience on the part of the designer, in terms of having encountered more rooms, and being able to determine while you're working with pen and paper where those travel-path errors will occur, so that when the design is implemented you have accomplished what you set out to achieve — a broad, accurate soundfield across the monitoring area of the control room." ■■■

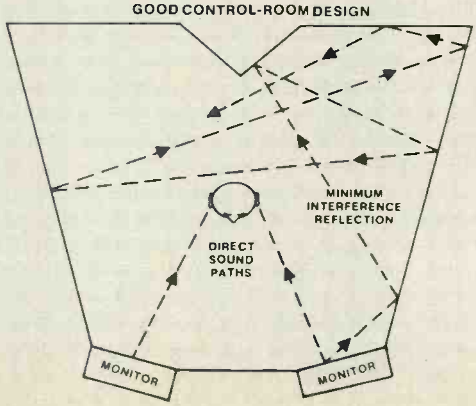
Room length and width are identical and parallel, producing continuous repetition until sound decays, and a maximum null at the resonant frequency in the center of the room. Speakers in corner produce HF travel path errors, and LF assist by reducing the radiating space.

Room length and width are varying, thereby producing a broad series of resonances, and thus a diffuse soundfield. Also, travel path errors are minimized.

BAD CONTROL-ROOM DESIGN



GOOD CONTROL-ROOM DESIGN



neer or producer be after that kind of 'larger' sound. The sound characteristic would be classified as medium to bright, depending on the setting of the two electrically operated drape systems — which can be drawn back to expose bare brick walls along the length of one side of the studio — and the position of the reflective panels along the other."

Several large panels, stretching almost from floor to ceiling, and covered with hard reflective wood paneling on one side, with thick pile carpeting on the other, can be swung out from the wall to create a mixed array of live or sound-absorbent surfaces. Waist-height gobos, also treated on each side with wood and carpet paneling, are available for local acoustic control around individual instruments or vocal sections.

"The studio has standard active trapping — in other words, enclosures within the walls and ceiling containing hanging acoustic blankets — but is not overly trapped. This allows the middle and upper frequencies to be kept at a higher level for the musicians playing in the room, to better hear their own instruments, and to improve communications across the area.

"Care was taken when calculating the room geometry — in particular, the various angles of sound reflection and refraction around the room — to keep the soundfield diffuse and true down to the longest wavelengths. The room has 'open and airy' sound which, of course, is a subjective description of the medium to long — 10 to 100 millisecond — spacing of the reflections, and the multiplicity of them."

The studio floor features three large hardwood sections that cover approximately 40% of the total working area, the remainder being covered in carpet over pad; the entire floor, like the control room, is laid on a conventional floating concrete slab design for sound isolation.

As in the control room, versatile adjustable lighting has been incorporated into the studio design, and is somewhat unusual in offering no less than three types of illumination.

"When warm lighting is required," Phoenix explains, "there are three zones of dimmable incandescent lights. Next, comes two levels of florescent work lights, used most commonly when something needs scrutinizing, such as while making instrument or miking adjustments, running repairs, or for reading music charts. Florescent lighting isn't generally considered as 'aesthetic' as incandescent lighting, and often is described as cold or harsh. But it is more efficient, both in terms of light/heat ratio, and power consumption.

"Lastly, and most unusually, the studio, like the control room, has an eight-by four-foot skylight in the center of the roof, which we inherited with the building; rather than close it off, we decided to make the skylight a unique feature of the environment. And if the musicians prefer not to work in natural lighting — either sunlight or moonlight — then

Studio C's control room is dominated by a large 4- by 8-foot skylight.

the skylight can be closed with an electrically operated shade that totally eliminates outside light when closed."

Video and Film Facilities

Studio C is well equipped with a variety of video and film screening facilities to handle an increasing number of scoring and sweetening dates. All patch panels in the control room and most of the other areas around the studio complex are fitted with video inputs and outputs, so that monitors, cameras, and video recorders can be set up wherever they may be required. A 25-inch color monitor has been mounted permanently in a strategic position between the control-room monitor loudspeakers, and can be switched to display a live or off-tape video picture, or console automation levels during audio-only sessions. A Panasonic NV9600 professional U-Matic VCR is on hand to replay video work tapes, while it appears to be no idle boast that the studio's BTX Shadow synchronizer can lock up virtually any combination of video and audio transports.

In the studio area, an electrically controlled seven- by 12-foot projection screen can be lowered from the ceiling for film-scoring sessions. (A projection room is to be found on the second floor, directly above the entrance to Studio C.) A pair of three-way Westlake TM-1 monitor speakers mounted either side of the projection screen provide studio audio playback.

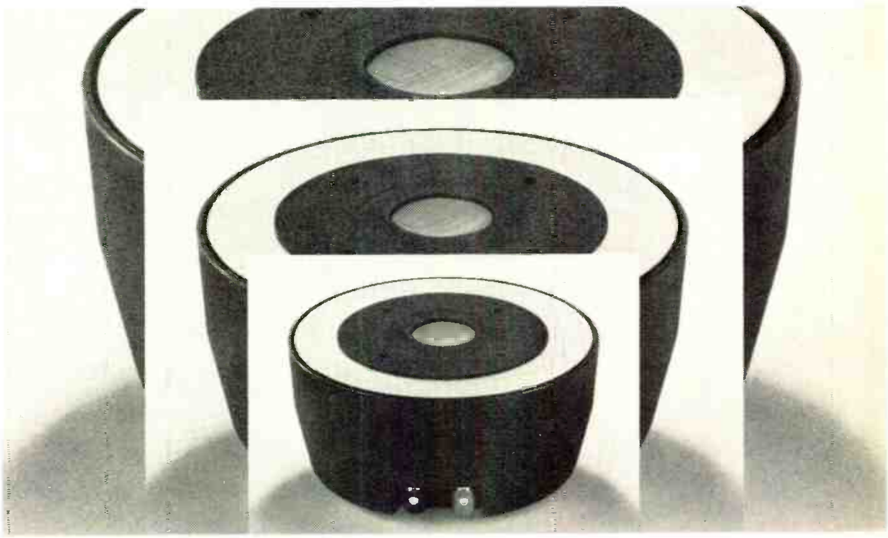
Additional Features

Designed to cater to just about every need of a producer, Studio C has several additional rooms or areas intended to make life as comfortable as possible during long protracted sessions. Two rooms, in particular, both with direct access to the control room have been put at the disposal of the hard-working production team. The entire 'complex' is purposely laid out to serve as a separate, practically self-contained studio area, which can be accessed through its own street entrance, and isolated as far as possible from traffic in the rest of the building during lockout sessions.

A "Producer's Refuge" — in reality, an over-sized soundlock between the control room and studio area — measures approximately 80 square feet, and can double as an overdub/isolation room, or private lounge. It features a wetbar, refrigerator, microwave oven, telephone, television, mike- and cue-line patchbay, plus direct access to its own restroom. A second area, known as the "Production Office," equipped with a desk and telephone, is intended to serve as an operations office for the producer, artist, and production team. It can be reached from the rear of the control room via its own private entrance.

* * *

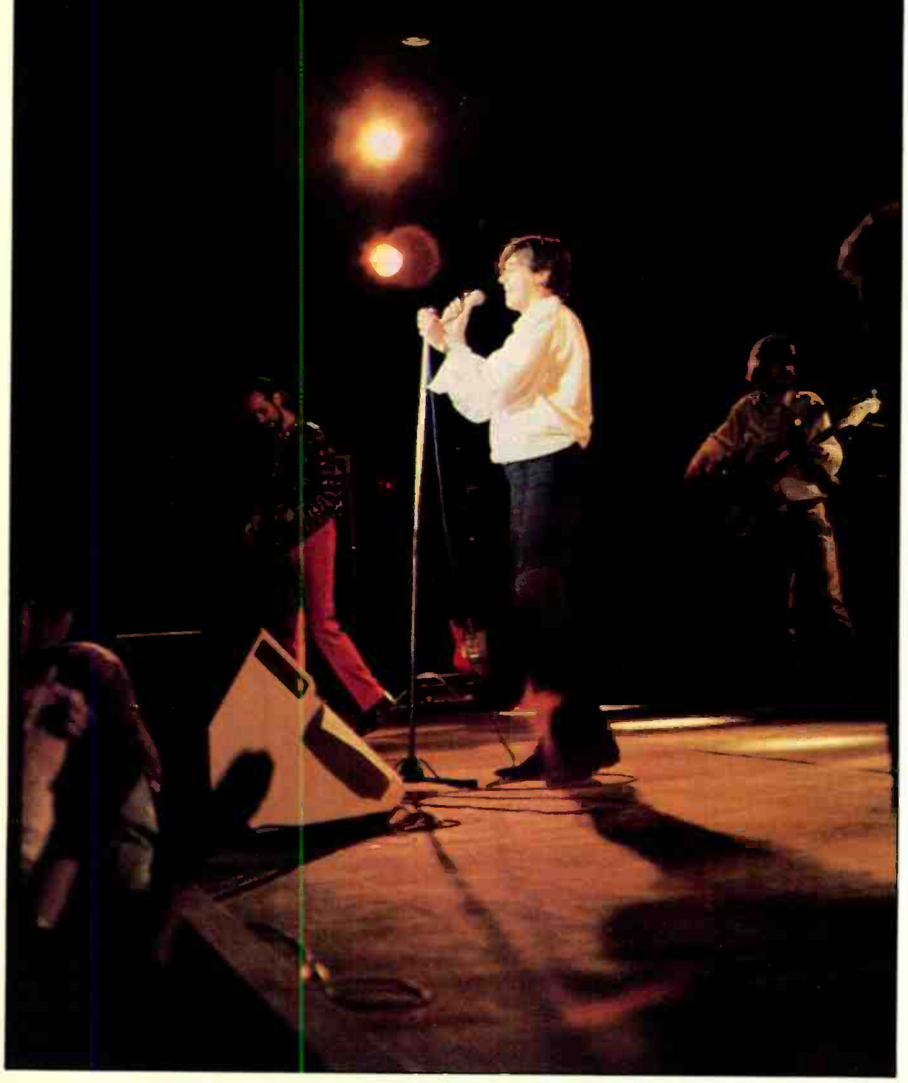
In terms of original design concepts, and the acknowledgement that Westlake readily makes to ongoing acoustics theories — including the Live-End/Dead-End approach pioneered by Chips Davis and Syn-Aud-Con, along with innovative designs from other leading acousticians — the new Studio C complex gives ever appearance of offering an accurate and creative recording environment, and one in keeping with today's increasingly complex and demanding multitrack sessions. ■■■



ELECTROTEC SOUND SYSTEM FOR ROXY MUSIC

New Lab-Q Loudspeakers in Operation at San Diego's State University Amphitheater

by David Scheirman



All photography by DAVID SCHEIRMAN

Roxy Music has not toured in the U.S. since the late Seventies. The band actually dissolved in 1979, and only came together again during early 1982 to produce *Avalon*; in April of this year, the group was on the road to promote that album, as well as the recently released, live four-song "mini-LP," *The High Road*. Twenty six shows were scheduled over a period of 30 days, in various venues around the country ranging in size from 6,000- to 16,000-seat capacity. According to Roxy Music's production manager, Chris Adamson, those 26 shows were very important to the group as it attempted to re-acquaint concert-going audiences with its music. Electrotec Productions, Inc. (formerly TFA) was selected to handle sound reinforcement for the tour, as well as supply stage lighting. Of particular interest was the fact that Electrotec's sound system for Roxy Music included the recently completed Lab-Q™ series loudspeaker cabinets.

Regarding the group's choice of sound companies, Adamson says, "It was basically a mutual decision made at management level. Robin Fox, our house engineer, felt that the new Electrotec system would be a good vehicle with which to present the group's new material, and we were particularly interested in finding a company which could offer a package deal on sound and lighting."

As Robin Fox recalls, "At rehearsals, I used four pairs of the new speakers in the main room at Studio Instrument Rentals [Hollywood], and I was favorably impressed with them indoors. This tour has a lot of indoor dates, but the outdoor shows are where we will really tell whether or not a system has what it takes. So far, I think we made a good choice."

Lab-Q Loudspeaker System

The Electrotec Lab-Q is a three-way active/fourth-way passive, two-cabinet speaker system, capable of being stacked on the ground, or flown in the air with no additional special hardware. According to Pierre D'Astugues, Electrotec's senior vice president, the new system is the result of more than three years of research and development. "The low-end of the system was computer-designed and manually tuned so as to offer a response to below 40 Hz," he says. "Our mid-cabinet is also computer-designed, and can achieve a true 60-degree coverage angle. Right now, the system is at a certain point in its evolution; this is not necessarily where we plan to stop with it. The basic cabinet structuring will be a constant, but as new transducer and electronics technologies hit the market we plan to adapt this system to fit the new developments."

D'Astugues feels that the Lab-Q system is fully taking advantage of today's available technology, but that a 10% to 15% improvement in the system will be evident by next year, due to engineering advances. "Feedback from engineers — [both] our own, and those employed by the accounts we service — is bringing about positive changes, and this system is exciting in that respect," he remarks. "It is an ongoing thing. Crossover circuitry, transducer manufacturing technologies, and component alignment are all three areas where advances will occur."

System Low-End

The Lab-Q low-end cabinet, of which 26 were used on the Roxy Music tour, houses a single 18-inch cone driver capable of handling 600 watts RMS. "The speaker is not currently available to the public," explains D'Astugues. "It has been developed especially for us by JBL, and is a bit different from the similar, commercially available speaker." When pressed for details, all he would offer by way of elaborating was that, "it's more expensive, I'll tell you that much!"

The cabinet, which covers the sound spectrum up to 250 Hz, is a folded-horn design with center bracing, and currently is equipped with two round porthole vents (Figure 1).

System Mids and Highs

The second half of the loudspeaker system is a cabinet of identical dimensions designed to stack directly via locking corner mechanisms on top of the low-end cabinet, and thus create columns that tower approximately 8 feet above ground level. The mid/high cabinet contains two 12-inch JBL E120s loaded in a deep horn chamber; a new constant-directivity, bi-radial horn mounted on a JBL 2445 driver; and two specially modified JBL ultra-high frequency units that are passively crossed-over (Figure 2). The system crossover

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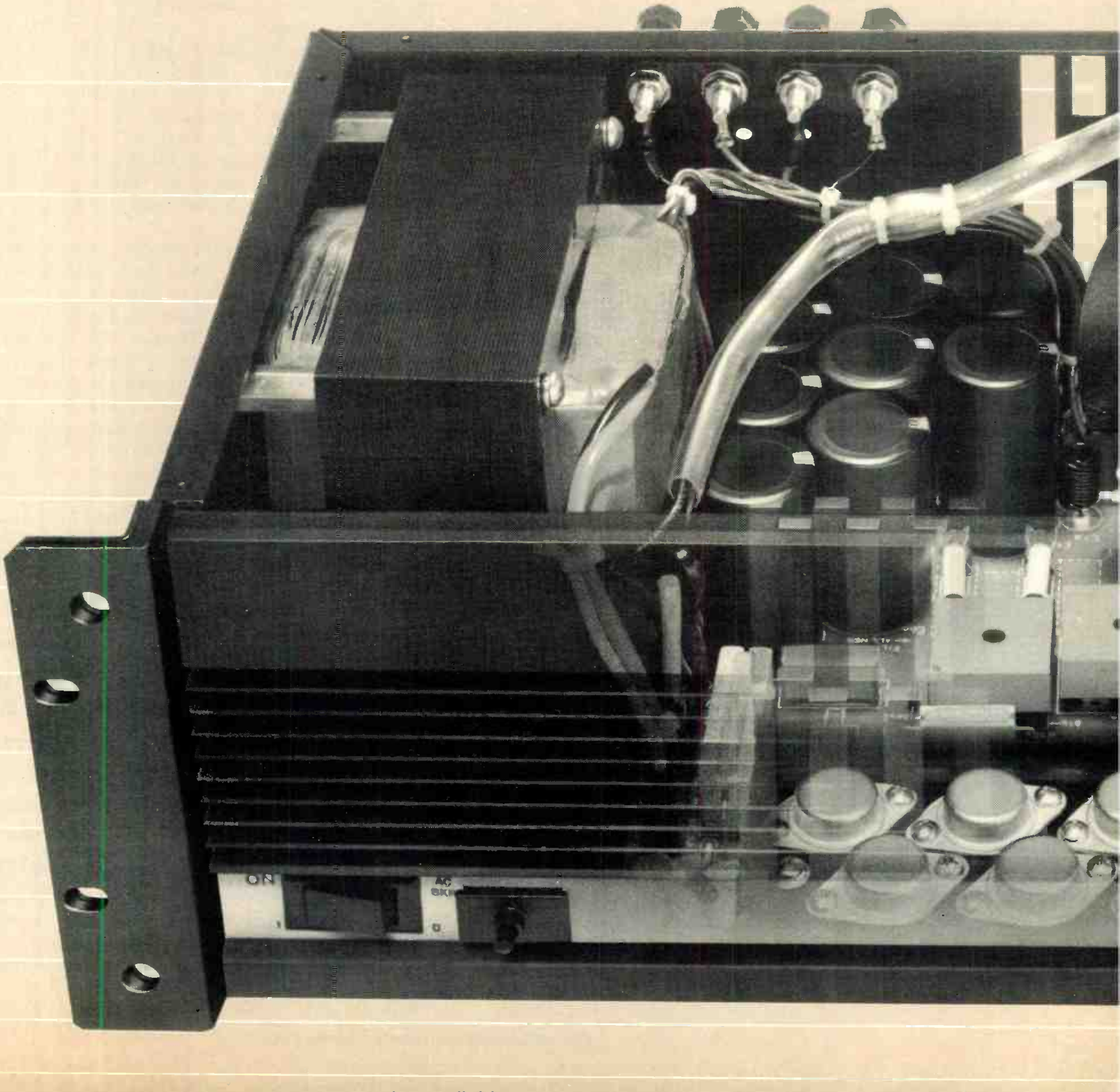
*Manufacturer's suggested retail price.



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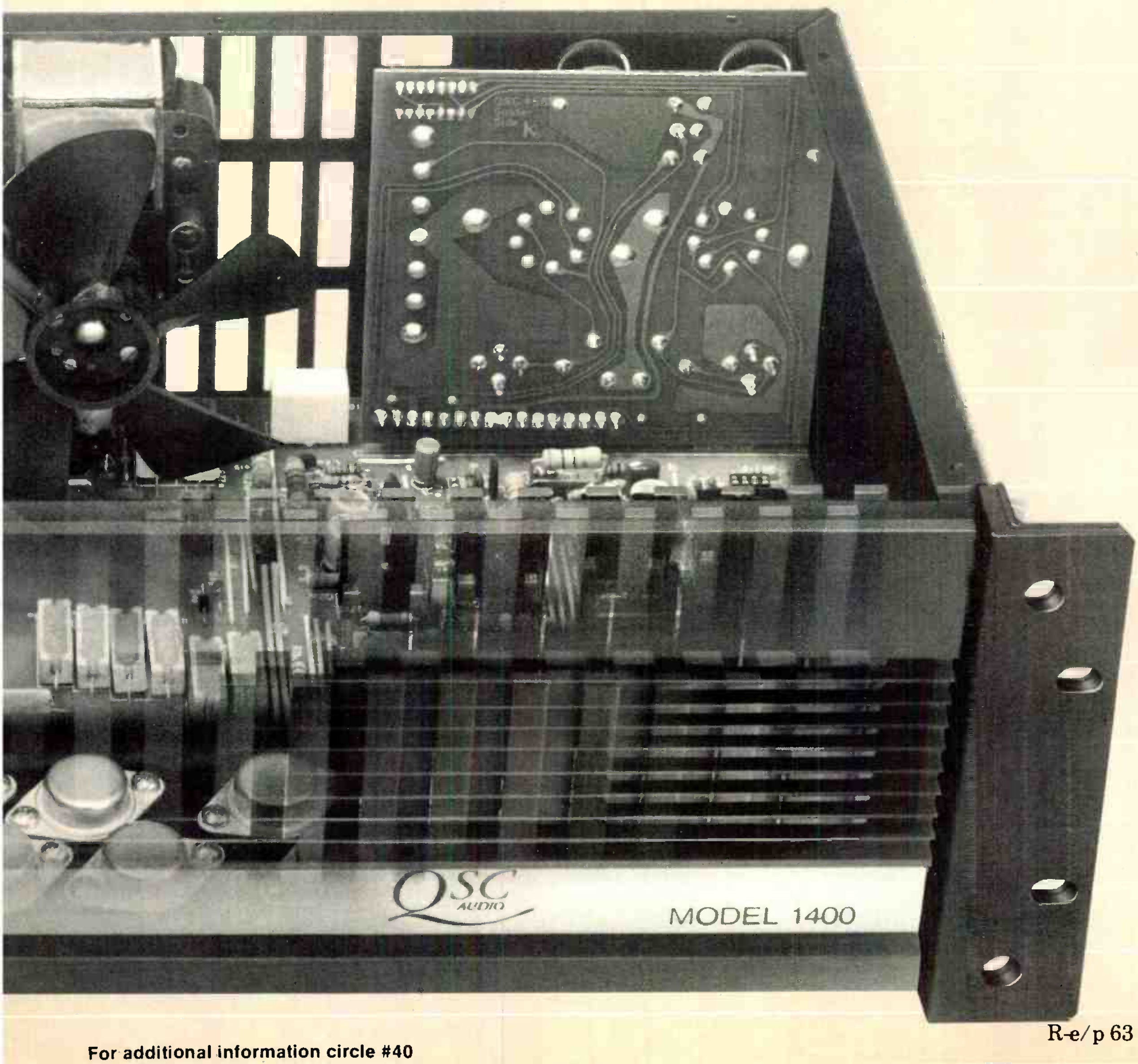

For additional information circle #40

provided Fox with an accurate display of the system's frequency response and sound-pressure level. An AKG C451E microphone supplied signal input to the analyzer, while an internal pink-noise source is available for system testing.

Other signal processing equipment in the house system included dbx Model 160 limiters for the main left and right outputs, and another pair of the same device for patching into the kick drum and bass guitar channels. Four Audio + Design Scamp Model 100 noise-gates were patched into drum input channels, and four Scamp SO1 compressors were available for processing vocals. Special effects devices included a Lexicon Model 224 digital reverb, Marshall Time Modulator, Audio + Design Compex F760X-RS stereo limiter, Eventide H949 Harmonizer, and an AMS Model DMX 15-80S stereo digital delay. House processing gear was contained in three identically sized 19-inch rack-mount cases (Figure 5), and was tied into the house mix console via dedicated multipair cabling.

In the house system, playback of prerecorded music was accomplished with two ReVox two-track reel-to-reel decks, and an Akai stereo cassette player.

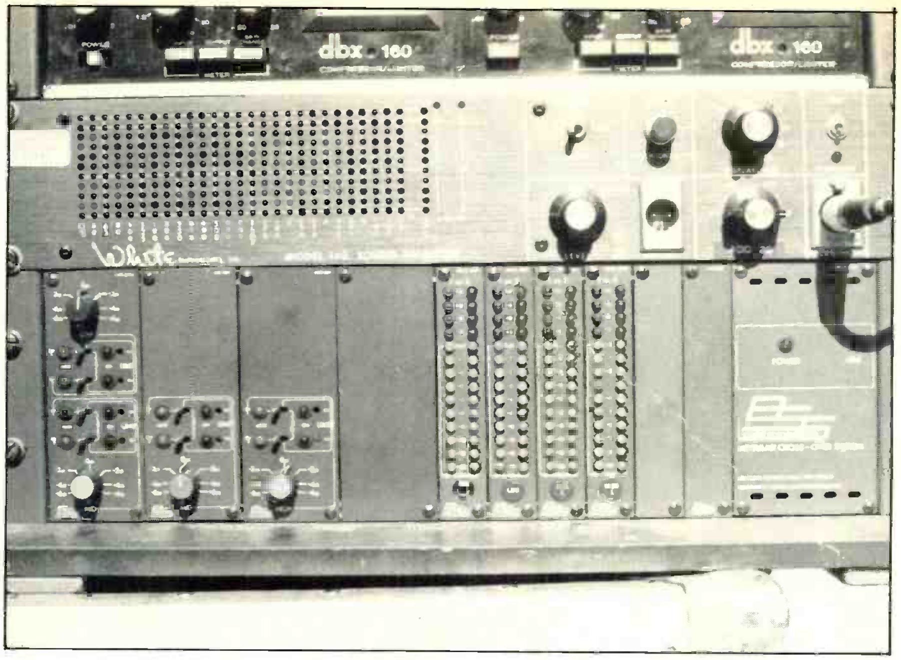

Figure 4: A Brooke-Siren system modular crossover with system points at 250 Hz and 1.5 kHz features built-in limiter circuits on each band.

Power Amplifiers

Electrotec has chosen the JBL Model 6233 as its exclusive power amplifier for both house and monitor systems. The Model 6233, because of its high-frequency switching power supply, weighs only 34½ pounds per unit, yet develops 400 watts RMS per channel into a 4-ohm load. A total of 63 of these amplifiers were on the road with Roxy Music, packed three to a case: 18 for the monitor system, and 45 in the house. Each amplifier rack has front-panel access to inputs and outputs, and AC connectors for each rack located at the base of the front panel (Figure 6). The 6233 amplifier contains one possibly unique design feature: a front-panel foam-lined vent provides flow-through ventilation to cover cooling requirements, so that units may be stacked directly one on top of another to conserve rack space with no ill effects.

System Set-Up

The concert that this writer observed was staged at San Diego State University's outdoor amphitheater, an open-air stage below ground level with high-rise

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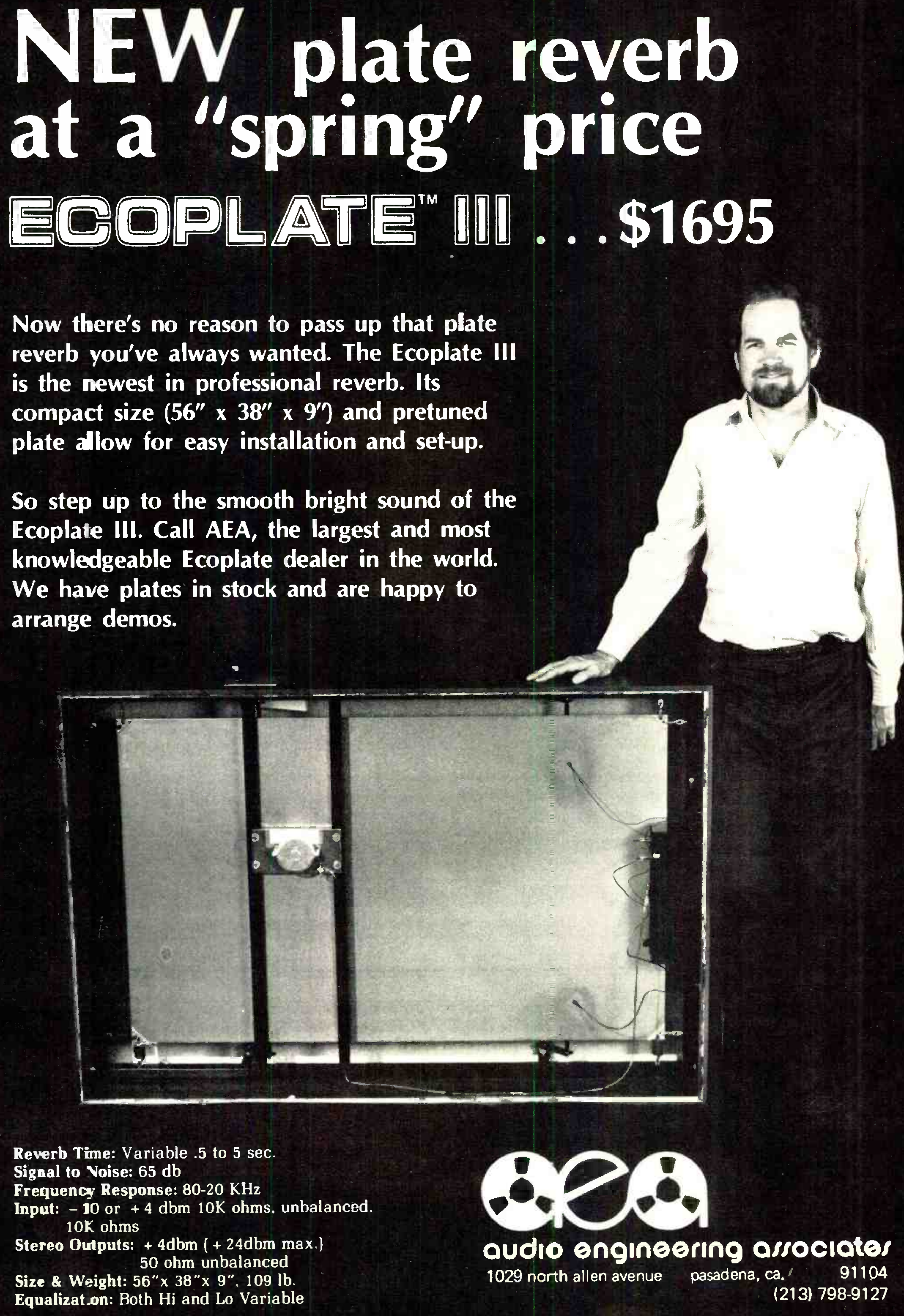
Figure 5: House mix electronics racks with comprehensive patching access includes a White RTA.



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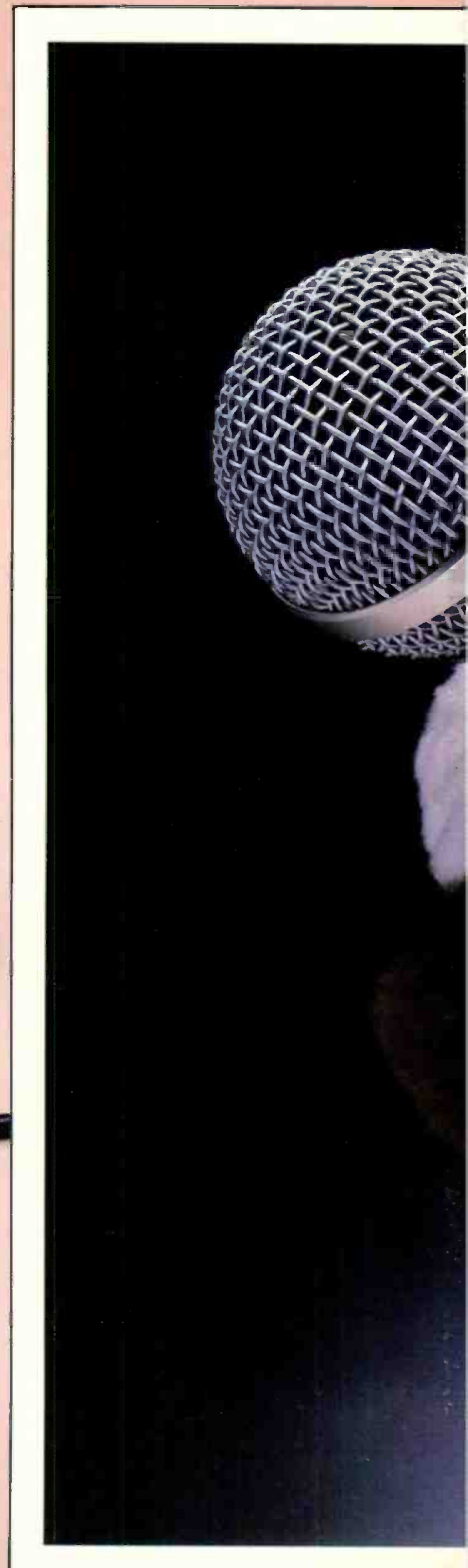

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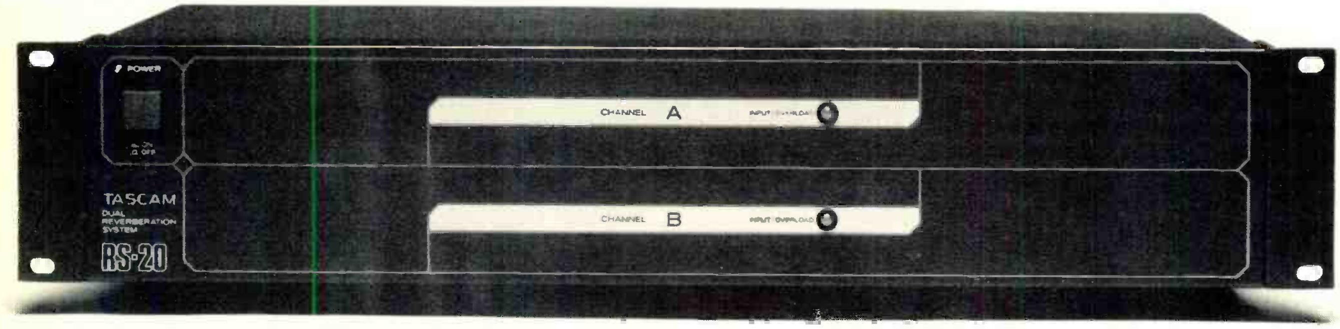

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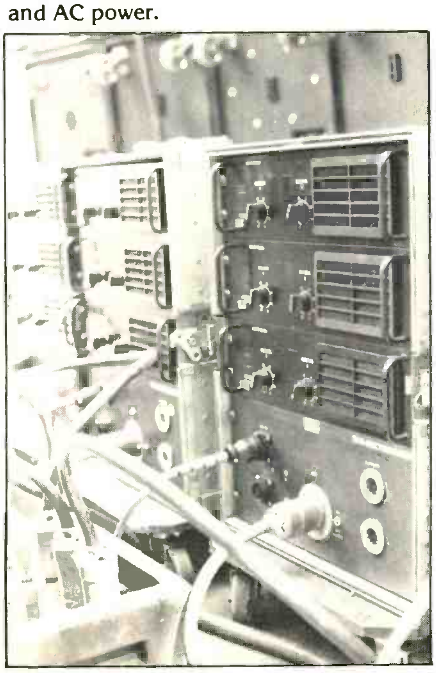
concrete seating that describes an area of nearly 140 degrees. To adequately cover such a venue, the main speaker stacks must be laid out to provide unusually wide coverage. Electrotec sound system engineer Mike Gibney chose to set up the system in two stacks of 21 boxes each. The stacks were given a slight hemispherical curve out to the sides, and in towards the center of house (Figure 7). Low-end cabinets were arranged in columns of three, and mid/high cabinets put up in similar fashion. The center of each stack comprised two columns of three low-end boxes apiece —quite a concentration of low-frequency energy in one spot. Since the distance between stacks was in excess of 60 feet, the monitor sidefill stacks were called upon to do double duty as center-house fill. Sidefills were placed slightly upstage of the front mike line, and angled across the downstage performing area, with their coverage pattern hitting the center house seating sections.

The San Diego concert was only the second show of the Roxy Music tour. Since the system was fresh out on the road from rehearsals, much time was spent during the afternoon before sound check carrying out basic road preparation: cables were labeled, case packing charts prepared, and loose hardware tightened. According to Gibney, basic system maintenance never ends when a system is on tour. "Every day there are things to watch for, gear to check on," he says. "For instance, right now I am tracking down a hum in the house system, which is not normally present with this gear. There seems to be a ground loop occurring at the interface between the house and monitor consoles [splitter box], and it is a trial-and-error process to

Figure 6: JBL Model 6233 power amplifiers are packed three to a rack, with front-panel access to inputs, outputs, and AC power.



DEVELOPMENT OF LAB-Q LOUDSPEAKER SYSTEM
— A Cooperative Effect Between JBL and Electrotec Productions

Assembling a new-generation concert sound system can be likened to building a race car: sometimes the parts you need are just not available off-the-shelf. Specific hardware needs for items such as custom-made transducers oftentimes are fulfilled only at great expense. However, when JBL was approached by Electrotec Productions for assistance in developing speaker cones and high-frequency units for the new Lab-Q loudspeaker system, the company was happy to help.

"We offer product development assistance for our major PA company customers," explains JBL's Mark Gander. "When a company is ready to place an order for a significant number of pieces — usually in excess of 100 — then it can become cost-effective for us to do a run of speakers or drivers which are slightly different from our normal product line. Usually, this will just mean putting out a certain driver in a 4-ohm version, when we normally do not offer it, or perhaps matching up particular components in a horn/driver configuration which does not appear in our product line.

"It's no secret that we have done special-order runs for most of the major sound companies: Clair, Maryland, Showco, and so forth. As long as the order is in sufficient quantity, we will do what we can to give them the specific components that they feel they need."

Pierre D'Astugues of Electrotec recalls that the Lab-Q system required both high- and low-frequency components which were slightly different than the commercially available units in JBL's line. "As a speaker manufacturer, JBL is very much oriented towards helping bring about new developments in the PA field," he says. "I feel that companies such as JBL like to use the concert sound companies to break new ground; to try out new products as they look into the future. What the PA companies do today, the rest of the business — the huge musical-equipment market — will do tomorrow. And, of course, with any new product there has to be a suffficient demand for a manufacturer to become involved.

"When we went to JBL, I told them straightaway that I needed at least 250, maybe 300, of the particular 18-inch speaker we were after. So, I think it was worth their while. Now, orders such as this have a positive effect on the industry in a two-fold way. First, it keeps a good relationship going between the manufacturers and the sound companies. Second, it benefits everybody else, because new products get off the drawing board and into the marketplace for all of the rest of us."

. . . continued overleaf —

DEVELOPMENT OF LAB-Q SYSTEM ... continued —

The low-frequency section of the Lab-Q loudspeaker system was designed with computer assistance. "JBL has computer programs in existence which help determine the optimum design features of transducers and cabinetry," D'Astugues explains. "We then entered our design parameters into those programs: the size of the cabinet; the conditions under which it would be used; the expected response . . . that sort of thing. And the box came out pretty much right on the money. Then we tuned the box manually; adjusting the ports, that sort of thing. We did that by listening to it, of course, using a variety of different criteria before finalizing the design."

JBL's Mark Gander once again stresses that such cooperation between builder and user is not unusual. "The past decade has seen many such design projects for us with a variety of customers," he points out. "And this sort of thing does bring about a gradual advancement of sound system technologies. It used to be that people just threw up stacks and stacks of big black boxes, assembled almost at random. Now we are working with some of our PA-company customers not only on proper design of the system loudspeaker components, but also on correct usage; proper placement of speaker cabinets in mathematically-correct arced arrays makes a tremendous difference in the sound system's phasing characteristics. And, because of it, live sound is getting better."

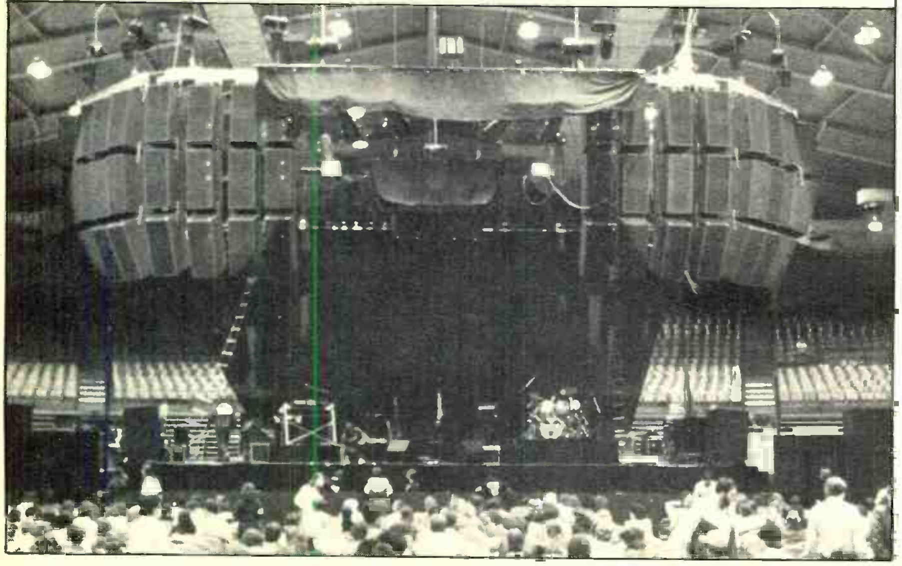
Where is PA speaker-system design going from here? "There are still some basic design parameters which will be with us for some time," Gander says. "Wood-pulp paper is still one of the best materials on earth for translating electrical energy into acoustical without too much coloration. The Japanese are experimenting with new materials such as polyproplene for 'speaker cones, but these are generally only working for a full-frequency application. The typical large concert PA system has the signal split into four or five different bandwidths, each requiring a specific, well-tailored transducer to most effectively reproduce those frequencies."

For horn design, Gander feels that the material used is not so important as the mathematics of the horn: flare rate, dispersion pattern, etc. "The new materials such as graphite composites may sound slick," he concedes, "but really all that is important in a horn is that the material be able to hold the horn shape in a rigid manner when it's in use! People who feel that they can hear a difference between identically designed horns made of wood and metal are only hearing whether or not the horn's builder has done a good job of maintaining a rigid, symmetrical form."

One area that promises future advancements in the use of metal components. "We have been moving away from the phenolic-type diaphragms in our compression drivers," Gander remarks. "The berylium-titanium compounds have brought about a much better diaphragm: improved frequency response, and greater durability. And, in magnets, the ferrite materials currently in use do not degauss so easily as the AlNiCo [Aluminum/Nickel/Cobalt] compounds used a few years ago. However, the ferrite materials are perhaps 15% heavier. The little 'walkabout-type' stereo headphones use an extremely lightweight selenium-cobalt magent, but this material doesn't work for the larger transducers due to its poor heat-dissipation characteristics."

Computer-assisted design and new materials technology promise further advances in speaker-system components development in the near future. And, cooperation between hardware manufacturers and sound reinforcement companies will help to draw these new developments into the commercial marketplace, making more efficient electro-acoustical transducers available to the public for a variety of uses. ■■■

Full Lab-Q flown system for Bob Seger concert, Knoxville, April 1983.



isolate and correct the problem."

As Gibney got the main stacks on the air and hunted down the few inevitable bugs, Electrotec engineer Chris Amison tuned the system for the band's engineer, and then proceeded to start the stage input hookup.

Stage Miking

Roxy Music features vocalist Bryan Ferry and saxophonist Andy Mackay, both of whom worked the downstage area. Ferry's vocal mike was a Shure SM-78, while the sax was picked up with two inputs: an AKG D224E was fed into a small Yamaha self-contained 6-channel mixer, sweetened with a Roland Space Echo, and fed through Yamaha speaker cabinets, one of which was miked. Additionally, a wireless unit was taken as a direct input for a clean, unprocessed signal.

Electric guitar amps were miked with Shure SM-57s, while the bass guitar was taken as a direct input, as were the keyboards. A Yamaha 12-channel mixer received signal input from Guy Fletcher's two Roland Jupiter synthesizers, Solina string synthesizer, Korg organ, Yamaha CP-80 electric piano,

Figure 7: San Diego State University outdoor amphitheater was given stacks of speakers three high, with the stack set up in a slightly curved array. One pair of Lab-Q boxes in the center is used as a sidefill monitor.

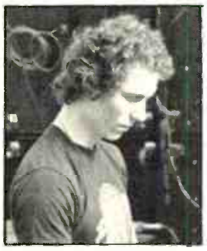
and Wurlitzer electric piano. The resultant mix was processed through a Roland SDE-2000 delay unit, and then sent to the house and monitor consoles. Additionally, a direct input was taken on each individual keyboard instrument to ensure clean signals at the desks.

Roxy Music's stage set this time out contained a generous amount of percussion, as can be seen from Figure 8. Andy Newmark's large drum set, complete with gong, was picked up with two overhead AKG C414EB condenser mikes. Toms were miked individually with Sennheiser MD421s, and the kick received an Electro-Voice RE-20. The snare was miked from the top with a Shure SM-57, and hi-hat cymbals given an AKG C451E. The auxiliary percussionists' congas, roto-tom, chimes, and bells were picked up with a pair of Beyer M160 mikes.

A three-way splitter with individual ground-lift switching on each input line passed the signal to the house and monitor consoles, leaving one split available for broadcast pickup or recording. Model 8226 AMP brand connectors on 11-pair multicable carried the microphone signals to the splitter box from satellite plug-in boxes distributed around the performing area. These boxes were designed with an extra female connector on the end, so that boxes could be "daisy-chained" for additional length when needed.

Monitor System

Stage monitors were handled by Electrotec engineer Bill Chrysler. The Electrotec Soundcraft 32-channel board was driving 13 separate stage monitor speakers, arranged on 10 mixes. "I'm using the 10 mixes for Roxy Music, so that still leaves me with six discrete outputs for opening acts," Chrysler offers. "After Roxy's sound check, I go around to the back of the board and change my output patching; that way the levels I've set for Roxy don't get touched until they come on stage again. Six more mixes is usually more than adequate for most opening acts."


BILL CHRYSLER

Chrysler's 10 monitor mixes for the headline band were set up as follows: sidefills; keyboardist; bass guitarist;

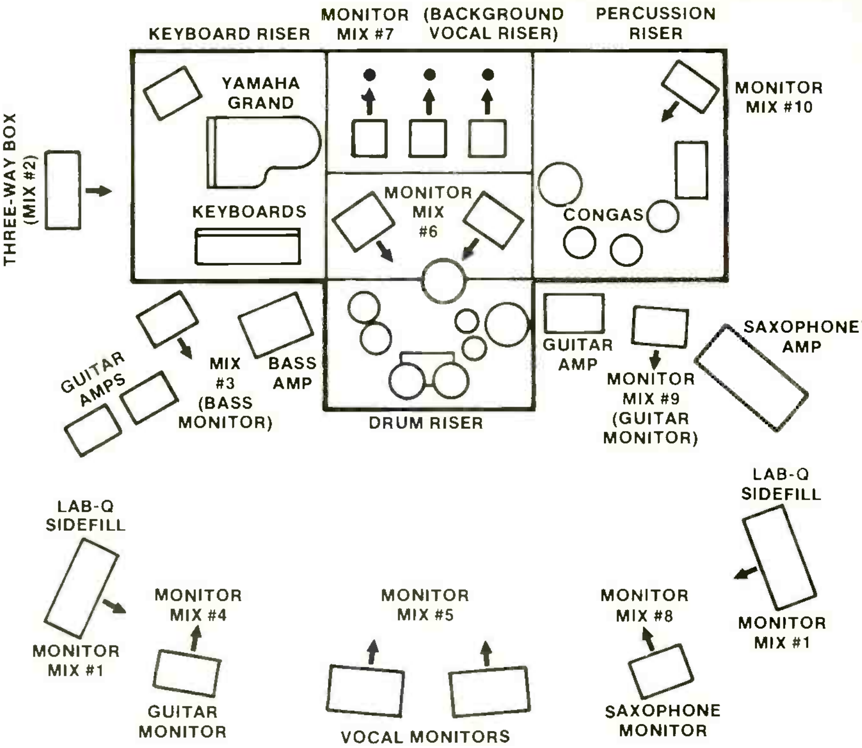


Figure 8: Stage and Monitor layout for Roxy Music Tour.

lead guitarist; Ferry's downstage vocal mix; drummer; background vocals; sax; stage-left guitarist; and percussionist. During sound check he was able to speak directly to each stage monitor mix area via the console's built-in talkback facility.

Monitor speakers were of three types. Sidefills comprised a Lab-Q stack on each side. The keyboard mix was heard through a three-way composite box measuring approximately 3 by 4 feet. All remaining mixes were driven through an Electrotec single-15 floor slant, which contains a single 15-inch JBL E130 speaker and a 2441 driver on a 2390 horn/lens attachment (Figure 9). The wooden box has a cutout around the lens that serves as protection for the fragile metal lens plates, while offering unobstructed lateral dispersion at the same time; it also serves as a useful carrying handle.

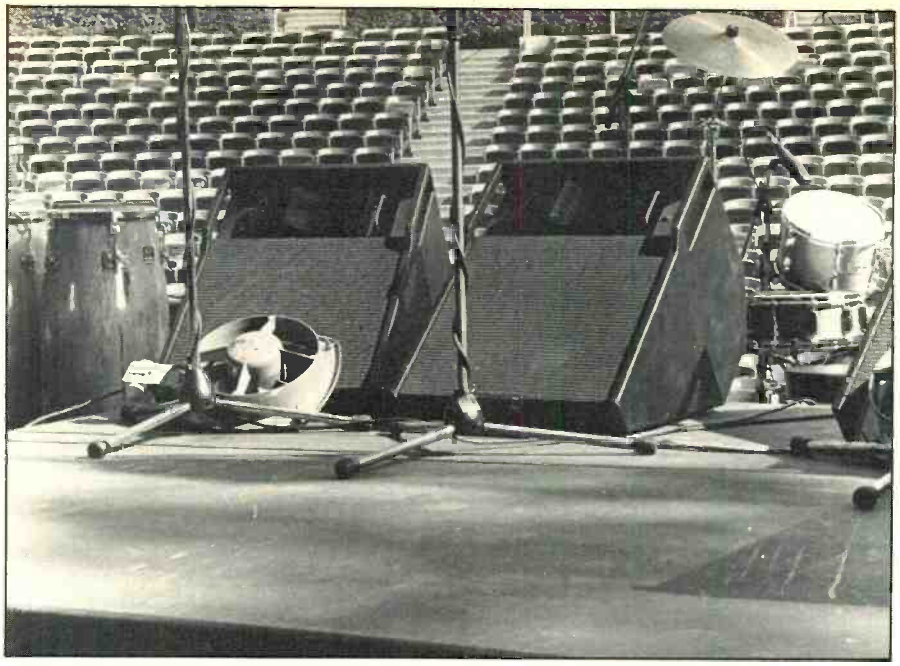


Figure 9: Electrotec single-15 floor slant monitor cabinets, as used by the background vocal section. Electric fans also were provided for visual effects.

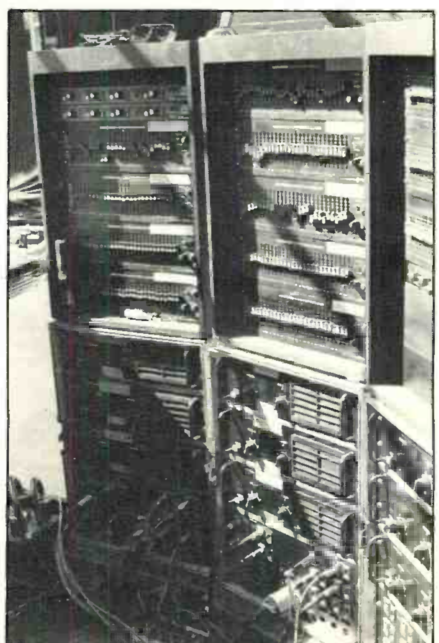
Monitor Electronics

Electrotec's monitor system features Klark-Teknik DN27 third-octave graphic equalizers, into which have been built the crossover units. Using the DN27's power supply, each custom-designed card offers bi-amp mix capability with a crossover point of 1.5 kHz. High and low output adjustments are located on the equalizer's front panel. The system Chrysler was tending for Roxy Music offered these third-octave EQ/crossover units on 12 of the board's available 16 mixes. Graphics and other monitor processing gear were contained in three electronics racks, which he stacked on top of his monitor amp racks for quick access (Figure 10). The racks included eight sides of Omnicraft GT-4 noise gates for drum use; according to Chrysler, however, they had not yet been necessary with this show.

One of the system's more sensible features was a spare power supply for the monitor console located in the same road case as the primary supply, prewired and switchable on-line should the need arise.

Referring to the monitor system at his disposal, Chrysler says that usually he will adjust the graphic equalizers during Roxy Music's sound check, and then leave them set. "I don't get into changing things just for an opening act — that can jeopardize Roxy's set," he continues. "If a mix does not sound right for the opening act with that EQ in-line, I can easily switch it out. Our wedges

Figure 10: Monitor electronics racks feature Klark-Teknik DN37 equalizers with built-in crossover units.



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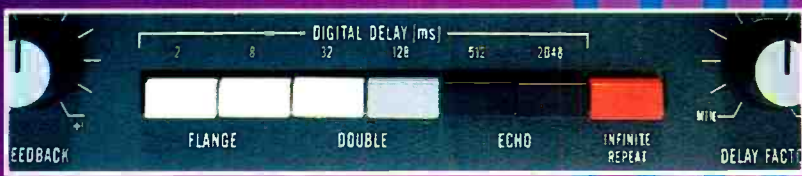

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quality," because it's a whole different ballgame recording with digital, where the quality is undisputed, as opposed to analog recording with all its well-known shortcomings. Good quality is not the same as a "good sound," right?

How does a kick drum *really* sound? Most of the record-buying public has never heard a bass drum without tape saturation. In Germany, 90% of record buyers have never even heard a bass drum that was recorded without a Kepex in circuit. Even *I* don't remember what a bass sounds like without modulation noise. What I am trying to prove, I suppose, is that rock music is — and should be — tape saturated!

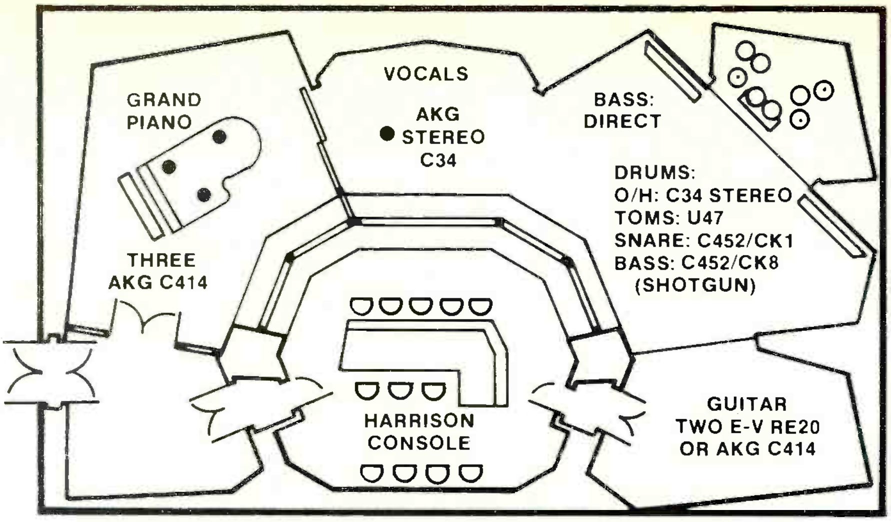
For the first couple of sessions for *The Visitors*, the first digital multitrack album to be produced in Sweden, I was carried away by the sheer sound quality coming out of the control-room speakers, and didn't care much for the *sound* of the instruments. Every engineer has experienced the painful situation when the band comes into the control room to hear a playback. The drums are always down 6 dBs after tape, and it's a whole different mix you're hearing. And you always say: "You guys should have heard it before it went on tape. Absolutely the greatest sound I've ever heard!"

With digital, however, what goes in, really does come out — no noise, no tape saturation... no nothing. Everybody on the session is impressed, including you. And you start to think that you can get away with anything. All that hard work on analog to preserve the sound after tape is no longer needed. You can start to live the life of a producer, you think.

What happens when you get home and listen to the 7½ IPS copy? You get the digital hangover. Those 90 dBs of dynamic range mean that you'll actually loose a lot of the music on your 30-dB home system. Things that normally you would have compressed or limited to keep over the noise level on analog tape vanish into the background. When the bass player hits the strings with a knuckle, it becomes *very* loud. The kick drum may vary 10 dBs, so you set your level on the loudest peaks, and then it disappears on your car stereo.

Lack of tape noise is actually another problem, for the analog man, anyway. Every engineer is familiar with the psycho-acoustic mystery that occurs when you're splicing leader tape to a fade-out ending of a song. You push the stop button when the fade is over, and you can swear that the music didn't stop until then. But you're always wrong — the music *always* stops two turns earlier on the reel. You're actually hearing things that aren't there. In other words, the sound lingers in your ears, long after it's gone.

This effect doesn't happen in the same way with digital, however; when it's quiet, it's *off*. Manufacturers of digital reverberation actually enhance the last portion of the decaying echo envelope with white noise, to overcome such an effect. The ear doesn't regard this as white noise, but rather that the sound continuity is undisturbed, as it would be with a natural echo.



STUDIO LAYOUT AND ROOM UTILIZATION FOR ABBA SESSIONS AT POLAR STUDIOS, SWEDEN. ENGINEER: MICHAEL TRETOW

On the 3M 32-track master tape, the noise in fact was lower than the actual "live" noise from the studio. Air conditioning rumble and mike pre-amp noise that was below the 87 dB digital threshold simply wasn't recorded. So, when you play back the tape for the musicians, you turn your monitors up a bit as you would normally do when working with analog. When the tape starts to roll, there's none of that familiar hiss, so you turn the monitors up some more. The . . . *Boom* . . . *Kaputt* goes your 'speakers.

Microphone Selection

Another thing you have to change when working with digital is your miking technique. Most of the time you choose mikes in respect to what they sound like on tape, *not* what they sound like in real time. You know that the peak at 20 Hz you hear when you're setting up the bass drum won't be there after tape, so why bother to mike it? In fact, since you're so used to what's going to happen on tape, you don't even hear it any more.

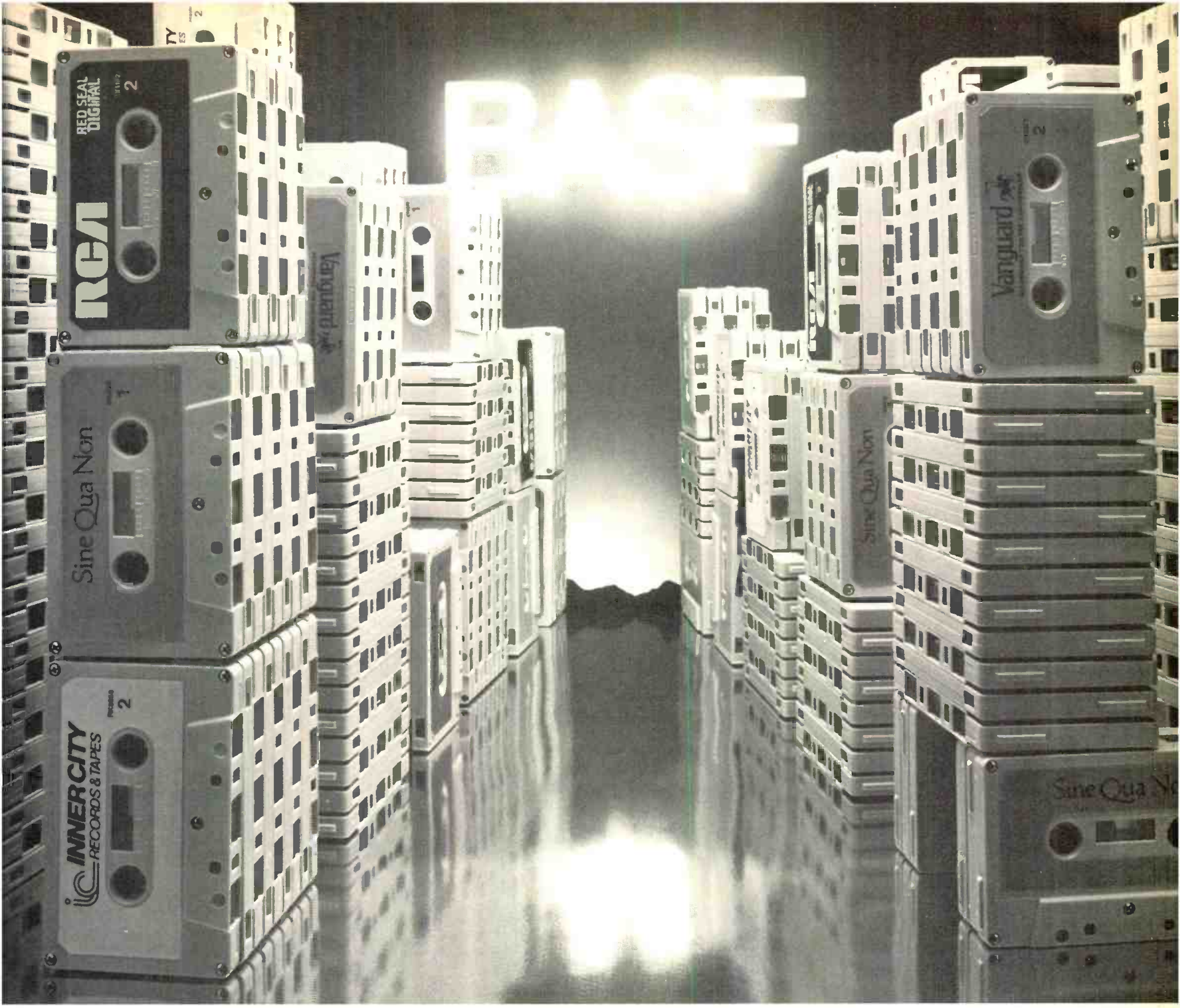
A few years back, I used to work in a room with a sharp peak in its frequency response at 40 Hz, and another at 80 Hz. After two years in that room I couldn't hear the peak any longer, and didn't understand what people were complaining about. I guess it's the same phenomenon when you live close to the railroad track; after a while you no longer hear the trains. The brain prefers not to take any notice of the sound even if the SPL makes your ears wiggle.

The 3M digital system picks up everything that is within the microphone's frequency response, and nothing is flattened out by tape saturation. Mikes you thought you could trust start to sound different. It's much like the switchover from the old tube boards to solid-state consoles. In those days, people believed in mirror impedances, which meant that the mikes were loaded with 200 or 600 ohms. When solid-state boards arrived on the scene, they loaded the mikes with 1.5 kohms, and all dynamic mikes started to change character. Even some condenser mikes sounded different when connected to solid-state pre-amps. I remember what the old U-47 sounded like on a tube board; it was nothing like the "warm" quality claimed today. On some singers it had a "shrieky" quality on particular syllables, which was just impossible to remove. Of course, there were no EQ sections on the boards in those days —just one knob marked "piercing," and one marked "dull." (And we only had the Fairchild Conax de-esser; remember the black boxes that substituted syllables with distortion?)

The 56-input Harrison desk in Polar's Studio A contains a number of transformerless mike pre-amps. For stereo piano I run two Neumann U-47s directly from the mike pre-amp output to the inputs of the 3M 32-track recorder, which gives a "warm" quality to the sound. Since there's no EQ involved, the sound becomes a little "muddy," so I always place a third mike close to the piano, add the EQ to that mike, and pan it to the middle on a separate track.

Polar Studios features an enormous glass-coated room that's just ideal for our Bosendorfer grand piano. Two glass doors seal off the glass room from the rest of the studio, so it's possible to place the mikes as far away from the piano as you prefer. The room also is great for strings; it can accommodate some 20 musicians, with full separation from the rest of the band.

On the song "Let The Music Speak," I used another medium-sized, all-wood room. The choir parts were recorded with very distant miking, so that sound reflections from the walls hit the mike with almost the same level as the direct sound. It gives a special, "boxy" quality to background vocals — very similar to a theatre stage recording, a "Broadway-type" sound that I like very much.



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Recording areas at Polar Studios . . .

. . . with variable acoustic environments

With five acoustically different rooms or recording areas surrounding studio A's control room, it is possible to achieve a different sound character by not placing the mikes too close. For the song "I Let the Music Speak," I used one room for the background choir parts, another room for the harmony vocals, and still another room for the lead parts. That way you can separate the different voices, since the ear detects the different environments as if it wasn't the same four people singing all of the vocal parts.

On vocals I mostly use an AKG C34 stereo mike but, due to the confusion mentioned earlier (with all mikes giving a new, unfamiliar sound on digital), I ended up trying a lot of different types, a decision that I deeply regret today when I listen to the finished record.

I try to avoid limiting or compressing vocal harmony mikes, but sometimes you're forced to. With lead vocals, I have a method I've worked out with the C34. Since it is a dual-capsule stereo mike, you can turn both capsules in the same direction. If you balance the two capsules on the board to give an equal output, and then compress just one of them, the unlimited capsule will take over on loud passages, thus minimizing the "strained" character you often get with limited voices.

Variety of Room Sounds

By using different rooms for different overdubs I try to achieve some depth perspective in the recording. I'm not very fond of recordings on which everything comes from left, right, or phantom center, with equal distance from the mikes. If something is supposed to be in the backgroud, it should be recorded with a background sound to it. Nobody records strings with close mikes, because distant miking has proved to give the richest, most musical sound.

Working with Abba it's very much a question of making room for all the overdubs. There can be some four or five different background vocals happening at the same time, three to four piano parts here and there, and maybe eight different synthesizer lines, together with the drums, bass and doubled guitars. If I was to record them all with close mikes, it wouldn't work in the final mix. Close miking results in a lot more transients that force you to keep down the mixing levels. If the congas are close to the mike, for example, it creates enormous peaks, with lots of energy on the track. Now, if the kick drum and piano both coincide on the same attack as the conga, the transients will be added and make those light meters flash like mad, although the apparent level from the monitors is no more than if the instruments were playing separately. If you move back a little with the conga mikes, they may even sound better because the room might add a little "body" to the sound. This also tells the listener that the conga is a *separate* percussion sound, and not a sound coming from the regular drum kit.

I always pick up the guitar with one close and one distant mike — not to get a "room" sound, but to let the guitar amp fill the room before the sound is picked up by the distant mike. I believe that the amp has to "breath" in order to produce a good guitar sound in the control room. Most of the time, I give the room mike a little vibrato from a DDL to make it "sing" a little.

I'll make use of different microphones because I don't believe in one mike offering absolutely peak performance on ever guitar amp, or any other instrument for that matter. Sometimes an Electro-Voice RE-20 does the job better than AKG 414 on guitar. It all depends on what the song is, what range the instrument is playing in, what amplifier is being used, and so on.

I once visited a very famous American studio that had a Sennheisser MD412

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microphone. I asked what they used it for, and they said vocals. You really use a low-priced microphone like that on vocals, I asked? The MD412 has been around since the Sixties (I've used it to record speech on 7½ IPS), and costs around $50 in Sweden. Nobody would *dare* to record vocals with a mike like that back home.

The first thing I did when I returned to Sweden was to try an MD412 on vocals for "When All is Said and Done." It was absolutely great! Never in a million years would I have guessed that such a simple mike could sound as clear and punchy as that. Which all goes to show that you have to try, and try again.

Abba keyboardist Benny Andersson owns a Yamaha GX-1, the "Dream Machine," a 16-voice synthesizer capable of producing some extremely rich sounds. On *The Visitors* album, I recorded the GX-1 in a different way from previous albums. Rather than use the instrument's direct outputs, I routed them through the console's foldback system, and out to the studio speakers. Polar is equipped with Philips Motional Feedback speakers with built-in power amps to cover all the different rooms. I miked each of the speakers in stereo — some of them very distant — and recorded the sum of them on two tape tracks, without any direct injection.

This technique proved to be very successful. It had none of that "electrical" sound to it that is common with synthesized strings. The string and brass sound was extremely natural, in fact, and even hard to tell from the real thing. On some cuts I used a Publison DHM89B2 Pitch Shifter for octave doubling on the most distant mikes; on slow lines it actually works very well, and gives a very "airy" sound.

Digital Editing

Rather than reaching for an Editall splicing block, to edit a digital album recorded on the 3M System you have to transfer each mix to a new reel of digital tape. To eliminate the count-in at the beginning of each song, it is necessary to record a few turns of silence before the song, since you can't splice leader tape to a tape. Then you rewind both machines and electronically edit out the

count-in. Once the edit point has been selected, the 3M editing system will perform the edit automatically. It sounds rather complicated, compared to simply splicing a piece of leader between songs, but it works just fine once you get used to it. One drawback, however, is that you can't change the sequence of the songs afterwards, without doing the whole album side over again. For *The Visitors*, I first transferred all the mixes to regular 15 IPS tape, and we then fooled around with the copies until we found the right sequence.

The cross-fade editing technique proved to be very useful on two-track masters, since there's no definite edit point that can cause abrupt changes in the stereo image. If fact, it was possible to edit between two takes of different tempos by moving the edit point to a "wrong" spot, and fooling our ears. Also, we did a lot of substituting on some of the songs. If the second refrain turned out better in the mix, we substituted the first refrain with the second, and so on. We mostly work with a click track of some kind, like the Linn Drum Synthesizer, but on songs with tempo changes and ritardandos, etc., a click track won't work. But, even if the tempos did not match at all, it was always possible to edit between the different parts of the song, by simply moving the edit point.

For the disk cutting stage, we had to wheel over the 3M digital machine to the cutting room, where Peter Strindberg cut the parts, by-passing the regular equipment rack, and feeding the cutter head amp directly from the machine's output; a 3M delay unit was used to provide the preview signal. Because we've had some weird experiences previously when each country did its own cutting, the album cut had to be repeated some 30 times by poor Mr. Strindberg and his crew, so that each country got their own, original parts.

By way of an example of how an unsupervised cut can go wrong, I once received a record of foreign origin, where the cutting engineer had compressed the *Volez-Vous* album a good 20 dBs. Strangely enough, this 3-dB dynamic range album sold more copies in that country than the previous albums had! *I* must have done something wrong, again. ■■■

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Greg Silsby talks about the New Sentry® 500 studio monitor...

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STUDIO UPDATE

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Northeast:

☐ **ATLANTIC RECORDING STUDIOS** (New York City) has installed a second Audio Kinetics Q.Lock 3.10 synchronizer for use with its Studer A800, MCI JH-24 multitracks, and Sony BVU-800 VCRs. "The capability of locking up our machines, in any combination, has greatly improved our ability to handle 48-track and video dates," according to studio manager **Paul Sloman**. *1841 Broadway, New York, NY 10023. (212) 484-6484.*

☐ **CRYSTAL CITY TAPE DUPLICATORS** (Huntington, New York) has expanded to larger facilities which feature a new Otari DP 7000 high-speed bin loop duplication system. The firm specializes in computer software duplication, in addition to dubbing music. Production output will be more than doubled by the new system. *48 Stewart Avenue, Huntington, NY 11743. (516) 421-0222.*

☐ **SOUND WAVE RECORDING STUDIOS** (New York City) has moved its disk cutting facilities to the following address: *2 West 45th Street, New York, NY 10019. (212) 730-7366.*

☐ **MIDNIGHT MODULATION** (Saugerties, New York) has upgraded from 8- to 16-track by installing a Tascam 85-16B interfaced with an Otari MX-5050B two-track. The outboard line-up now features Lexicon digital delay, UREI and dbx compressor/limiters, and an Eventide H949 Harmonizer. New Neumann and AKG condenser microphones have been added the the studio's collection. Instruments include a Yamaha grand piano, a Korg Poly 61 synthesizer, and an ARP 2600. **Michael Bitterman** owns and operates the studio. *2211 Pine Lane, Saugerties, NY 12477. (914) 246-4761.*

☐ **TROD NOSSELL RECORDING STUDIOS** (Wallingsford, Connecticut) is now providing a new slide pulsing service for audio/video clients, providing pulses of any frequency to trigger projectors. A Linn Drum Computer is now offered on a rental basis, and an Altec 21B tube microphone has been added to the studio's mike collection. *Wallingsford, CT. (203) 269-4465.*

☐ **UNIQUE RECORDING STUDIOS** (New York City) has totally redesigned Studio B's control room and installed an MCI JH-528 console with producer's desk.According to studio manager **Joanne Georgio**, the facility can now accommodate larger groups and "in-control-room" multi-synthesizer recording. New equipment includes an 8-voice E-Mu Systems Emulator, a 6-voice MemoryMoog, and a Garfield Electronics Dr. Click drum machine/synthesizer/sequencer interface. *701 7th Avenue, New York, NY 10036. (212) 921-1711.*

☐ **THE SCHOOL OF AUDIO ARTS** (New York City), a division of The Center for Media Arts, has installed a large audio equipment package supplied by Martin Audio Video Corporation of New York. The equipment list is headed by an Otari MTR-90 Series II 24-track recorder and Otari MTR-10 2- and 4-track recorders. Also included in the package are 17 Otari MX-5050B two-tracks and an MX-5050 MKIII-4 four-track, an Eventide SP2016 Digital Processor, JBL Bi-radial monitors, plus Ramsa 8118 and 8112 mixers. **Harry Hirsch** is the dean of the Audio Arts division. *226 West 26th Street, New York, NY 10001. (212) 609-8870.*

☐ **RIGHT TRACK RECORDING** (New York City) has added an Audio Kinetics Q.Lock 3.10 SMPTE synchronizer to its equipment complement. A Solid State Logic console is interfaced to the Q.Lock which, in turn, controls Studer A800 and MCI JH-114 multitracks. Says studio owner **Simon Andrews**, "This combination makes synching a delight." *168 W48th Street, New York, NY 10036. (212) 944-5770.*

Southeast:

☐ **CRITERIA RECORDING STUDIOS** (Miami, Florida) has taken delivery of a Valley People Kepex/Gainbrain II noise gate/limiter package, a Lexicon 224X digital reverb unit, and an Audio Kinetics Q.Lock SMPTE synchronizer. All the new gear is intended for use by Criteria's five divisions: Studio Recording; Remote Recording; Disk Mastering; Film and Video Services; and Media Post-Production. The Media Post-Production facility is a new Criteria operation, designed to assist the commercial market with all aspects of audio and video production. Services include complete multitrack analog and digital recording; ¾-inch off-line video editing utilizing Sony BVU-800 VCRs, convergance editor, and time-base corrector; scoring to picture (film or video); and audio sweetening to video utilizing the Q.Lock system. Remote audio and video recording, dubbing, transfers, and laybacks also are offered. **Douglas Weyrick** is manager of the new division. *1755 NE 149th Street, Miami, FL 33181. (305) 947-5611.*

☐ **THE REX HUMBARD FOUNDATION** (Boynton Beach, Florida) has installed two MCI JH-24 multitracks, and an automated JH-600 Series console. The producer of evangelical television programs records between six and seven fully scored musical selections for each half-hour broadcast. SMPTE interlock is used for interface with video. The foundation's new 12,000 square foot facility will employ 60 people upon completion, and will mean a shift of Humbard's production from Akron, Ohio. **Donny Humbard** is the foundation's business administrator, while **Charles Humbard** is chief engineer. *Boynton Beach, FL. (305) 732-0220.*

☐ **CRESCENDO RECORDERS** (Atlanta, Georgia) has re-opened its Studio B after installing a new Soundcraft 2400 Series console, and a Studer A-80 Mark II 24-track. *125 Simpson Street North West, Atlanta, GA 30313. (404) 223-0108.*

South Central:

☐ **OMEGA AUDIO** Dallas, Texas) describes itself as the first studio in the region to offer 46-track recording capability, with the addition of a second Otari MTR-90 Series II 24-track. Also added is a Lexicon 224X digital reverb unit. The facility continues to maintain its 24/48-track remote recording facility, which has been kept busy with numerous television projects during recent months. *8036 Aviation Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

OMEGA AUDIO — second Otari MTR-90

☐ **STARGEM RECORDING STUDIO** (Nashville) has added a pair of dbx 900 Series outboard racks that include two 902 De-essers, four 903 Over-Easy compressers, a 907 stereo gated compressor slave, four 904 noise gates with PLM and key functions, three 905 equalizers, and two 906 Flanger/Doubler/delay lines. Additions to the studio's outboard rack include an F769X-R Vocal Stressor, a Loft 450 Flanger, a DeltaLab DL5 Harmonicomputer, URSA MAJOR 8X32 digital reverb, and an LP-1 Plate Reverb. *43 Music Square East, Nashville, TN 37203. (615) 244-1025.*

☐ **THE CASTLE RECORDING STUDIO** (Franklin, Tennessee) has installed a new Lexicon Model 97 Super Prime Time, and a new Lexicon 224X digital reverb system. These outboards complement the facility's Harrison 48-channel 3232C console, and two Studer A80VU recorders. Monitoring is handled by JBL 4330 and 4311s, with Auratones on the console. Mikes are by Neumann, AKG, Shure and Sennheiser, while the instrument package is headlined by a full-length Bosendorfer grand piano. Castle proprietor **Joseph Neyens** also announced the appointment of **Chuck Ainlay** to the position of chief engineer. *Old Hillsboro Road, Route 7, Franklin, TN 37064. (615) 791-0810.*

☐ **SOUNDS UNREEL STUDIOS** (Memphis, Tennessee) has upgraded from eight- to 24-track operation with the installation of a Sound Workshop Series 30 36-input console feeding an Otari MTR-90 Series II multitrack with Autolocator, and an Otari MTR-10 two-track. An URSA MAJOR 8×32 digital reverb also has been added to the outboard rack. *2027 Courtland Place, Memphis, TN 38104. (901) 276-8468.*

☐ **DIGITAL SERVICES** (Houston, Texas) has purchased a second Sony PCM-3324 24-track digital recorder, along with an Audio + Design Recording signal processing package, which includes an F769X-R Vocal Stressor and SCAMP S01 compressor/limiters, S05 dynamic noise filters, S100 noise gates, S23 equalizers, and S25 de-esser modules. *2001 Kirby, #1001, Houston, TX 77019. (713) 520-0201.*

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STUDIO UPDATE

FACILITIES
EQUIPMENT
PEOPLE

Mid-West:

□ **REELSOUND RECORDING** (Southfield, Michigan) has re-opened its doors after extensive remodeling and enlarging of its facility. Two new isolation rooms and a drum booth have been added, as well as a Neotek Series II 28/24 console, Tascam 85-16B 16-track, Sound Technologies Ecoplate reverb, Korg Poly 61, and assorted microphones. A Yamaha grand piano also has been purchased. *25859 Mulroy Drive, Southfield, MI 48034. (313) 356-2640.*

□ **TRC NORTH** (Indianapolis, Indiana) has opened its second 24-track recording studio, Studio B. TRC president **Gary Schatzlein** says that the new studio has been designed to complement the existing Studio A, located on North Illinois Street. While Studio A is equipped with a Harrison 3232 console, Studio B boasts an MCI JH-528 board. Valley People 65K Series automation is utilized in Studio A; the computer in Studio B is an MCI JH-50 system. While Studio A is described as being smaller and more intimate, Studio B is larger and boasts 14-foot ceilings and multiple isolation booths. Both rooms utilize MCI JH-24 multitrack and JH-110 mastering machines, although Studio B also boasts a Studer A-80. Both rooms use JBL 4435s as primary control monitors, but alternate monitoring systems are available in each. Main monitor amp is a Crown PSA-2 in Studio A, and Acoustat Trans-Nova Twin-200 in Studio B. Both control rooms were tuned by audio consultant **Jerry Milam**. Studio A utilizes a Lexicon 224 digital reverb unit, while B has an EMT 140 plate. Each studio also has AKG BX-20E echo, and both have multiple noise gates, limiters, delay lines, etc, an Aphex Aural Exciter, and a large complement of microphones. Schatzlein offers that the second studio was built "because demand for time in Studio A was forcing some clients to wait or to use inconvenient hours. Also, the suburban location is more convenient for some clients, while other continue to prefer downtown." *5761 Park Plaza Court, Indianapolis, IN 46220. (317) 845-1980.*

□ **WHITE HORSE RECORDING STUDIO** (Moline, Illinois) has purchased a new E-mu Systems Emulator digital keyboard, and a new Drumulator digital drum machine. The new instruments complement the studio's MCI 16-track and two-track recorders interfaced with a Tangent Series 16 console, a Revox PR99 two-track deck, Studio Technologies Ecoplate II reverb, DeltaLab DL2 Acousticomputer, dbx 900 Series rack, MXR Pitch Transposer, and Omnicraft GT-4 noise gates. Monitoring is handled by UREI 811, JBL 4311, and Auratone 5Cs. The microphone collection includes units from Neumann, Shure, AKG, Sennheiser, Beyer, Crown PZM, Shure, and Electro-Voice. White Horse also features a custom designed four-line cue system, allowing each musician separate studio headphone mixes. The studio is owned and operated by **Ron Spencer** and **Jeanne McKirchy-Spencer**. *620 15th Street, Moline, IL 61265. (309) 797-9898.*

Mountain:

□ **J.V.J. RECORDING STUDIO** (Flagstaff, Arizona) has upgraded its 8-track studio facility with the addition of a Tascam Model 15 16/8 console. J.V.J. has also acquired a MICMIX Master-Room XL-210 reverb chamber, an Orban 622-B parametric EQ, plus Altec-Lansing and Auratone Sound Cube monitors. **Julian, Viola,** and **Jay Diaz** are the studio owner/operators, with the latter acting as studio engineer. *22 Ridge Crest drive, Flagstaff, AZ 86001. (602) 774-8113.*

Southern California:

□ **AWAKENING PRODUCTIONS** (Los Angeles) have upgraded its studios with the installation of a Soundcraft 400B console supplied by New World Audio, Inc. of San Diego, CA. The company also delivered and installed a Tascam 85-16B 16-track machine, and an Eventide

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

SP2016 digital reverb unit. *930 North Wetherly Drive, Los Angeles, CA 90069. (213) 659-6238.*

□ **LION SHARE RECORDING STUDIOS** (Los Angeles) has equipped its control rooms with Auratone Model T6 Sub-Compact Two-Way systems. A pair of these monitors are to be found in each of the three control booths, while the studio also is utilizing a pair of the larger Model T66 Compact Two-Way Systems, and a set of the Model QC66 Three-Way Systems for recording, referencing, and mixing. *8255 Beverly Blvd., Los Angeles, CA 90048. (213) 658-5990.*

LION SHARE — Auratone monitors

□ **HIT SINGLE RECORDING SERVICES** (San Diego) has upgraded from 8- to 16-track with the installation of a Stephens 16-track on two-inch recorder equipped with 811D electronics. The deck operates at 15 or 30 IPS, with a Varispeed option for tape tuning and special effects. The outboard rack now includes a Valley People Stereo Dyna-Mite combined limiter, noise gate, ducking and keying device. A Lexicon Model 93 Prime-Time also is on order. *College Grove Center, Lower Court 4, San Diego, CA 92115. (714) 265-0524.*

□ **AUDIO CASSETTE DUPLICATOR COMPANY** (North Hollywood) has been opened by **Steve Mitchell** and **Steve Katz**, both former staff engineers at A&M Studios. The facility features real-time tape duplicaiton, and is equipped with a 3M M79 Series for producing 15 and 30 IPS masters, a TEAC 7030 for half- and quarter-track 7½ IPS masters, and 11 Sony TCK-777 cassette machines for the dupes. *North Hollywood, CA. (213) 762-2232.*

□ **NEW WORLD RECORDING STUDIO** (San Diego) has installed Otari MX-5050 MKIII-8 and MKIII-2 transports to supplement its Otari MTR-90 24/16-track and MTR-10 two-track decks. In addition, a BTX Shadow/Cypher video interlock system is being installed in the control room. *4877 Mercury Street, San Diego, CA 92111. (619) 569-7367 or (800) 854-2006.*

□ **CAPITOL STUDIOS** (Los Angeles) has acquired a Sony PCM-1610 digital system, according to **Charles Comelli**, manager of studio operations. The rig consists of two PCM-1610 16-bit digital audio processors, two companion BVU-800D U-Matic VTRs, and a DDU-1520 Digital Delay Line used in disk cutting. In addition to recording studio duties, the system will be used in preparation of Capitol's XDR audiophile cassette series. A Capitol staff engineer has been sent to Sony's school for digital training, and the label currently is said to be considering purchase of a Sony PCM-3324 multitrack machine. *1750 North Vine Street, Hollywood, CA 90028. (213) 462-6252.*

□ **INTERNATIONAL EDUCATIONAL RECORDINGS** (Del Mar) has contracted New World Audio, Inc., of San Diego, for the installation of a new Tascam 38 eight-track and a Tascam Model 50 mixing console, upgrading the operation to eight-track. *243 12th Street, Suite C, Del Mar, CA 92014. (619) 755-6419.*

Northern California:

□ **MUSIC ANNEX RECORDING STUDIOS** (Menlo Park) has completed the first phase of total reconstruction of its Studio C, according to owner **Keith Hatschek**. The original 30- by 20-foot room once served to augment the 24-track Studios A and B, and as an 8-track demo and rehearsal studio. Client needs soon put 24-track hardware in the tiny 10- by 12-foot control booth of Studio C. As a result Music Annex has cut through the wall to a 6,000-square-foot space next door, and built a new control booth for Studio C and for a 3,000-square-foot video stage being built in the newly acquired area. The old control room has been converted to an isolation booth.

Northern California . . . continued —

Acoustical design is being handled by **George Augspurger**, who also handled the recent redesign and re-equipping of Music Annex's Studio B. Studio C is now equipped with an AMEK 28/24 console, MCI JH-114 multitrack and two-track machines, and an outboard rack featuring a Lexicon 224 digital reverb with 4.4 software program, and an EMT stereo plate. UREI 811 monitors are mounted in a ceiling track that can be adjusted within an "acoustical window," allowing the mixer to optimize the stereo image to any position at the console. In addition, Gauss sub-woofers are flush mounted in cabinets at the front of the room. These are totally non-directional, handling the frequency spectrum from 80 Hz down. Augspurger carried out the minimum room tuning required with White third-octave EQ. The new booth will be used for album-oriented tracking, mixing, and video sweetening, marking a move away from "music-only" services. An Otari MX-5050 half-inch 8-track system also has been acquired to allow cost-conscious projects to be completed in the state-of-the-art facility. *970 O'Brien Drive, Menlo Park, CA 94025. (415) 328-8338.*

MUSIC ANNEX — new Studio B

☐ **THE SOUND SERVICE** (San Francisco) has added a Neotek Series II 24/18 mixing console with semi-parametric EQ on each channel; an MCI JH-110 Audio Layback System for C-Format one-inch videotape; a six-voice polyphonic sequencing synthesizer; and a digital drum machine. The MCI ALS deck is designed specifically to handle delicate videotape masters, and offers audio signal-to-noise figures said to be 6 dB better than video transports; wow and flutter specification is comparable with any studio-quality mastering machine. The new equipment will allow new services to be offered by the studio, including video syncing of film dailies, and tailor-made music search cassettes. Audio and video film post-production also are featured. **Steve Shapiro** is the studio's in-house synthesist. *860 2nd Street, San Francisco, CA 94107. (415) 433-3674.*

☐ **STUDIO C** (Stockton) is responding to the increased use of its New England Digital Synclavier digital synthesizer with the addition of engineer/synthesist **Ralph Stover** to the staff. *2220 Broadridge Way, Stockton, CA 95029. (209) 477-5130.*

☐ **INDEPENDENT SOUND** (San Francisco) has installed a new Otari MTR-90 24-track machine, to be interfaced with the facility's Sound Workshop Series 30 console; the board also has been upgraded with the addition of an ARMS tape-based automation system. *2032 Scott Street, San Francisco, CA 94115. (415) 929-8085.*

Canada:

☐ **COMFORT SOUND** (Toronto, Ontario) has expanded to 24-track with the purchase of a new Ampex MM-1200. *2033 Dufferin Street, Toronto, Ontario, Canada M6E 3R3. (416) 654-7411.*

AUDIO/VIDEO UPDATE

Eastern Activity:

☐ **SHEFFIELD RECORDINGS** (Phoenix, Maryland) finds **John Ariosa** and **John Palumbo** producing a video for the Criminal Records group, **Crack the Sky**. The piece, slated for MTV, will feature Penthouse centerfold **Divina Celeste**. *13816 Sunny Brook Road, Phoenix, MD 21131. (301) 628-7260.*

☐ **NATIONAL VIDEO CENTER** (New York City) has reached agreement with **ATI Video Enterprises** to supply video taping and post-production facilities for ATI's trio of music-oriented programs. *Night Flight* and *Radio 1990* air on the USA Cable Network, while *FM-TV* is syndicated in over 70 markets for broadcast television. Interviews have been taped with Neil Young, Lou Reed, Journey, Grace Jones, and Devo at National's TV-2. Special effects were incorporated into these segments via National's computer editing suites, utilizing the facility's DVE, ADO, and Chryon character generators. **Jeff Franklin** serves as executive producer, **Cynthia Friedland** is producer, and **Stuart Shapiro** produces and directs all three programs. In other developments at National, **Norman Kasow**, a specialist in music and sound effects for motion pictures and television for the past 30 years, has brought his extensive libraries to National's studios. Kasow's career spans work on *Car 54 Where Are You?*, Woody Allen's *Sleeper*, and the soon-to-be-released Mike Nichols' film *Silkwood*, as well as commercials for such agencies as Doyle, Dane, & Bernback, Leber-Katz, and J. Walter Thompson. His collection boasts over 45,000 sound effects arranged by categories from Airplanes and Animals, to Sports and Trains. A vast assortment of stock music rounds out the library. According to chief mixer **Dick Mack**, Kasow will couple his expertise in film and video sound mixing and editing with National's Audio Kinetics Q.Lock and Vidi-Mag systems in National's continuing effort to offer complete film and video services under the same roof. *460 West 42nd Street, New York, NY 10036. (212) 279-2000.*

WPCB TV 40 — new audio control room

☐ **WPCB TV 40** (Wall, Pennsylvania) has completed construction of a new audio control room to be utilized in live TV production. The room was designed by WPCB audio supervisor **Rick Shaw**, and features a Tascam Model 16 24/8 mixer bussed together with a Tascam Model 50 12/8. Ampex ATR-700 stereo machines handle mastering, while JBL 4430 monitors powered by Crown PS-400, -200, and D-75 amps are on hand for monitoring. A TFT modulation meter is located in the monitor cluster, and a custom designed 72 by 24 mike switching matrix has been installed, fed by units from Shure, Sennheiser, and AKG. Other sideboards include a UREI LN-1176 limiter, BiAmp reverberation, and a Symetrics Dual Gate. **Jerry Foreman** is the chief engineer, while **David Kelton** is the station's general manager. *Wall, PA 15148. (412) 824-3930.*

Central Activity:

☐ **SOUND ARTS RECORDING STUDIO** (Houston, Texas) has entered into the mobile recording world with its first multitrack audio/video shoot, which featured the **Barbara Pennington Band.** The piece was recorded live at the Rocksy in Houston, with final audio production mixed at Sound Arts' 24-track facility utilizing SMPTE timecode for sync with the video decks. Video was handled by Television Video Productions' **Alton Christensen** and **Steve Long**, while Sound Arts' engineers **Jeff Wells, Mark Nooney,** and **Charlie Cosme** handled the audio. *2036 Pasket, Suite A, Houston, TX 77092. (713) 688-8067.*

Western Activity:

☐ **THE COMPLEX** (Los Angeles) played host to **Van Halen** for the lensing of their *Top Secret* video project, produced by the band and Lombard Entertainment Company; the piece was premiered at the recent US Festival to an audience of 300,000 fans. Video facilities for the shoot were provided by Video-Pac Systems; sound was by Nova Sound Research; and lights were provided by Keylight. *2323 Corinth Street, West Los Angeles, CA 90064. (213) 477-1938.*

☐ **RUSSIAN HILL RECORDING** (San Francisco) has been completing work on post-production for the upcoming Disney feature film, *Never Cry Wolf*, directed by **Carrol Ballard**. Down the hall, **Bernie Krause** and **Gary Remal** are scoring *Vanished* for Home Box Office, with **Richard Greene** engineering, while a deal has been struck to commence post-production on the Ladd Company release of *The Right Stuff*, directed by **Phil Kaufman**.. **Richard hymns** will supervise dialog, while Greene engineers. *1520 Pacific Avenue, San Francisco, CA 94109. (415) 474-4520.*

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Our new 4312, for example, represents the next step in the evolution of the 4311. Improvements include a new high resolution dividing network for better transient response and a mirror-imaged design that provides enhanced stereo imaging. These refinements significantly improve the loudspeaker's performance, yet maintain the unique sound character that made it an industry standard. And best of all, the 4312 is still priced to fit comfortably in even modest budgets.

For those that require a more flexible or compact monitor, we've created the 4411 and 4401. These loudspeakers incorporate our most advanced component and design technologies. Both the 4401 and 4411 utilize newly developed transducers arranged in a tight cluster to provide outstanding coherency of sound for close monitoring. This design also minimizes off-axis variations in the far field. Additionally, the 4411s are mirror imaged for improved stereo perspective.

For maximum flexibility, the continuously variable levels controls on the 4411 are calibrated for both a flat direct-field response and a rising axial response that produces a flatter power response. And for ease of adjustment, each of the monitors' level controls are baffle mounted. Finally, the low frequency loading has been optimized for flat response when the speakers are placed away from room surfaces. Because of this, the 4401 and 4411 may be console mounted without the loss of low frequency response typical of other designs.

For additional technical data and a complete demonstration of the 4312, 4401, or 4411, contact your local JBL Professional Products dealer. And discover the next generation of compact monitors. From the refined to the redefined.

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For additional information circle #53

An Engineer's Guide To Music

Understanding the Basic Syntax of a Session — Part Two

by Jimmy Stewart

As we saw in the June issue of *R-e/p*, a musical score or chart can act as a useful "catalyst" for a recording session. Part One of this article considered the basic blueprints of a song, musical structure, writing your own music charts, and the myriad details than can be gleaned from a conductor's score. In this, the concluding part, we move on to consider the more complex form of a master rhythm chart, and the increased information to be found on a full music score. Also provided is a list of recommended recordings that can further underscore the importance landmarks to be found on music charts.

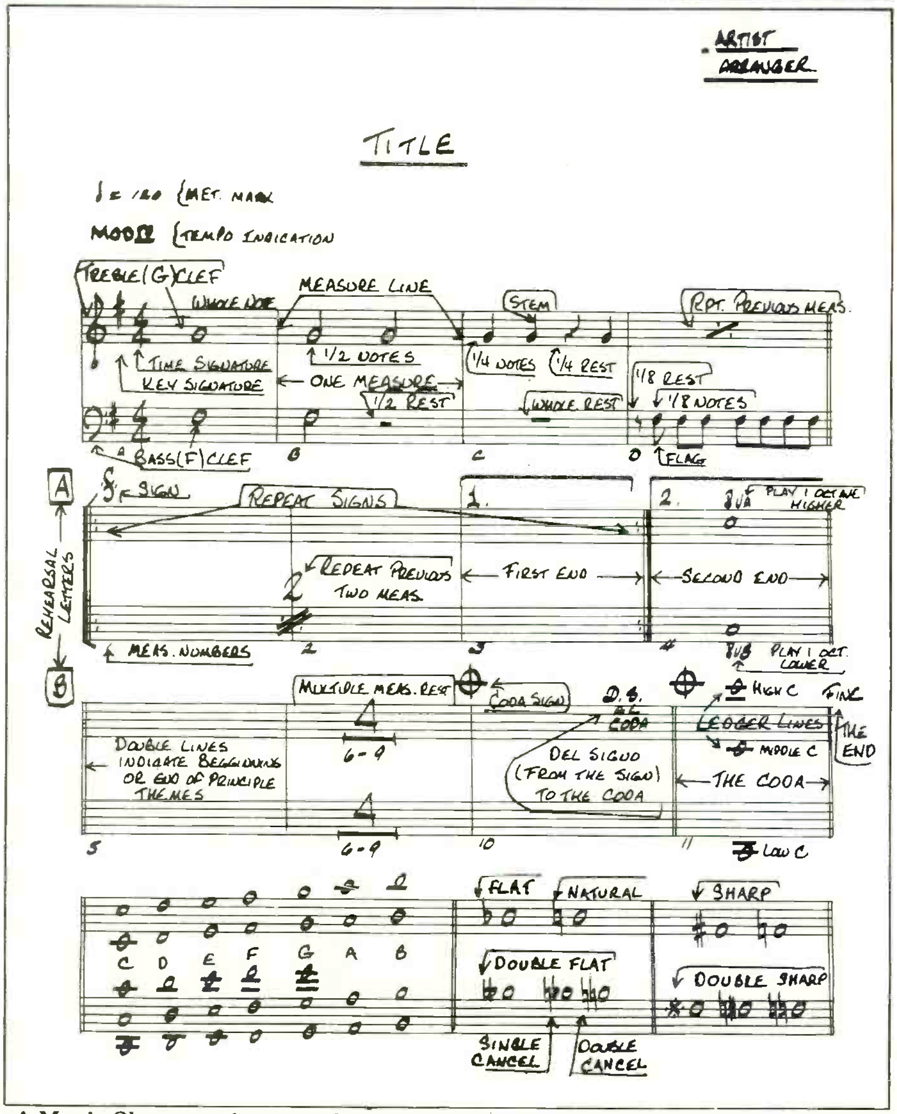
Master Rhythm Chart

Larry Brown is an engineer who also plays drums and piano. In his everyday work he encounters many different types of production situations, ranging from direct-to-disk sessions, to hours spent working with a singer and lead sheet in the multitrack process. Quite often he might have to work with a detailed Master Rhythm Chart, upon which all the essential elements will be notated.

Referring to the accompanying Rhythm Chart, Brown explains the process: "We have the bass tuned down to a low C. Without hearing the tune I might consider throwing a pitch shifter on the bass, only because of the low tuning. It's a good idea because there's space here in the breakdown [or way in which instruments enter the song].

"It looks as if the tune is going to move more into tempo in the A section; the movement also is there in the A section, and there's a guitar figure. The intro basically is a line that is now setting up more of a groove into the A section, leading on to the B section.

"In the C section it looks like another whole kind of feel setting itself up. C2 again is opening back up: it has a guitar playing whole notes, and the keyboard players playing whole notes, with the bass figure still moving. So, looking at that I'd say it's going to open up again. We have four quarter-notes marked at the repeat with slashs over them so the players can lay into it. Going back in you're going to be nailing the repeat section, which is repeating back into the A section.



A Music Sheet can be considered as a useful "roadmap" of a session. Shown above are some of the typical markings to be found on a score, including tempo, time and key signatures, repeats, and similar notations.

SVC-350

Vocoder

Incomparable Product

The Roland Vocoder is the first truly sophisticated Vocoder to be priced reasonably enough to be accessible for most applications. The Vocoder requires two inputs: the carrier input and the program input. The program input consists of spoken or sung words which are input through a microphone. The carrier usually consists of an instrument input such as a synthesizer.

The Vocoder circuit consists of two major sections: the analyzing section and the synthesizing section. The analyzing section breaks down the voice (or program) input to determine its frequency or content at any given instant, then re-assembles this sound in the synthesizer section using the instrument (or carrier) input as a basic source of building material. The program or voice input is analyzed by passing it through a set of eleven Voice Character Frequency filters.

The Mic (program) input features both XLR and phone connection jacks, and a Level control with LED indicators for optimum level settings. There are two carrier inputs, one for most instruments, and the other specifically for guitar, with a control designed to tailor the Vocoder's response to guitar harmonics. These carrier inputs both feature Level controls and LED indicators.

An interesting feature of the Roland Vocoder is a Hold feature that holds the Vocoder tone color as long as desired and is used to bridge gaps where the singer takes a breath. An Ensemble (or Chorus) effect serves to give added dimension to the Vocoder sound.

For more information contact RolandCorpUS, 2401 Saybrook Ave., Los Angeles, CA 90040.

SVC-350 Vocoder Specifications

Microphone Input (program)	1/4″ phone jack or XLR connect 600Ω, −54 dBm min.
Instrument Input (carrier)	1/4″ phone jack 100 kΩ 0 dBm max.
Guitar Input (carrier)	1/4″ phone jack 100 kΩ
Guitar Amp Output	1/4″ phone jack 5 kΩ
Mono/Stereo Output	1/4″ phone jacks
Power Consumption	8 watts
Dimensions	19″ (W) X 3.5″ (H) X 9.7″ (D)
Price	$995.00

For additional information circle #60

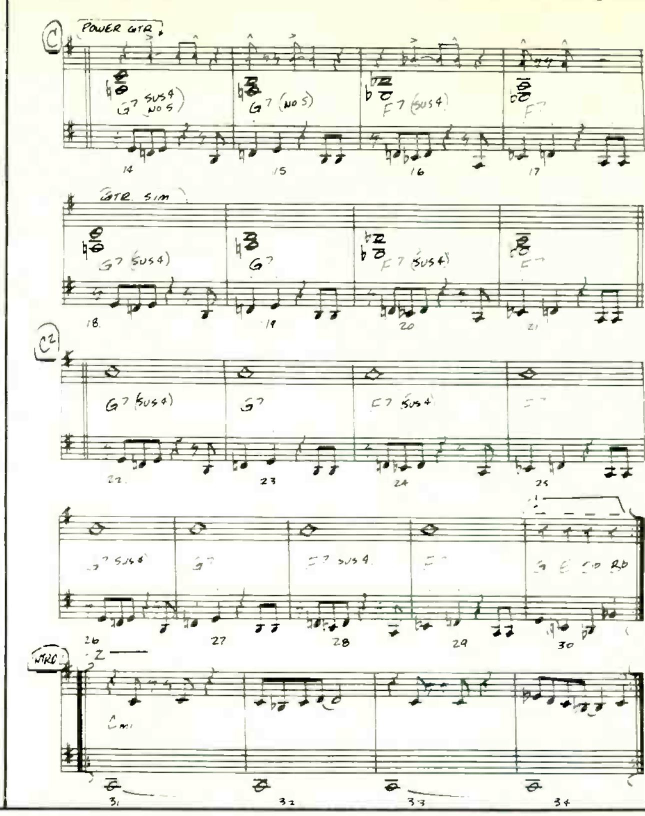
Roland

Incomparable Quality

BASIC RHYTHM CHART FOR "NIGHT LINE"

Page #2

"Night Line" **. . . continued — Page #3**



"I would do nothing to the faders at this point. In fact, I don't recommend engineers jumping all over faders in the case of a soft dynamic part where you're getting into a noise problem. The idea is not to get rid of your dynamics. Moving faders only helps you when you get into that noise-ceiling problem, and you don't want to get a lot of tape hiss. You don't want to destroy what the players are out there trying to put on tape, in the sense of covering up dynamics. All you're trying to do is help yourself in the mix situation, where all of a sudden you find the noise ceiling is louder than your program ceiling.

"The frequency spectrum on the chart looks pretty meat and potatoes — strong midrange — outside of the fact you've got this low-bass part happening that's tuning down to a C. The guitars are playing pretty much in the middle of their range, and it looks like there are some block-type chords. The keyboard would be pretty much in there with the guitar; basically you've got single lines written out. You might have keyboard or synthesizers double up on some of the lines since it would help to support it. Again the bass figure is heavily notated throughout, and it looks like it plays a pretty important role as far as the groove goes. I would spend a certain amount of time being sure that the bass sound is going to hold up, because it looks like it's pretty important.

"I see that some 'power' guitar chords are marked on the chart, even though there's no dynamics written. I'm going to assume at this point in the tune that the C section is going to take off somewhat, and open up a little more. I notice that the D section is going to the breakdown; the bass is basically holding a note out for two bars, and represents a moving guitar figure. I assume that the tune will loosen up a little bit there. That leads into the breakdown, which is basically starting bass and drums.

Looking at the chart, it repeats four times, and I'm guessing that the writer is adding the second or third time, even though it's not notated. The guitar probably joins in with it. Also, there's a modulation coming up into the fade, which looks like a breakdown section that's gonna keep spinning. Again it's a groove thing; I'd be sure I had plenty of tape, because sometimes a lot of the magic might happen on the 12th time it spins around. Even though you may want to cut some of it later, you need to be sure there's enough tape to catch it all.

"Although there's no dynamics written on the chart, you would assume they don't play a major role, since the arranger didn't notate them as such. It's probably more of a dance tune, or a straight ahead piece. The chart doesn't have that many ups and downs going for it.

"I would pick up the score first thing on the session, and look through to see what instrumentation there is. Of course, I would probably go out into the room and check it for myself, and ask the guitar player what kind of axe he's going to pick up — would it be acoustic or electric? — and what kind of sound he wants to shoot for."

Production Score Sheet

Larry Brown also may be asked to work from a production score for a large orchestra. Although he might not have time to interpret the music, Brown says that he would listen to the run down and mark dynamics on his production sheet, along with what else he though necessary to the session. By way of an example, he was asked to cast his eye over the accompanying chart for "Shadow of Your Smile."

"Dealing with it right off the bat I can see we've got a full orchestra, since there's string parts," he offers. "Starting at the top, the intro is a muted string figure. Because the strings are in mutes there are no dynamics, and they're playing pretty soft. At that point I would group my strings on the board, so that I could make some subtle dynamic changes here.

"The song starts with a three-bar string intro, the strings wear mutes, and the vocalist starts. He has a pickup into the fifth bar, which is market 'A Tempo,' so that's the first bar of the tune. I wouldn't mute the vocal mike, because he may want to do a little ad lib in front of his opening line. Into the seventh and eighth bar we have a harp glissando. That will affect me on the board, because I want to be sure I have the harp mike open. If you're dealing with a big orchestra — and working with a lot of TV soundtracks — many times you will be going in and out of tunes quickly. There may be an earlier tune on which you didn't have harp; I may have shut that mike down, and now need to open it.

"Going into the seventh bar there also is a horn figure written. On the 11th bar I have flutes entering again. At bar #13

Garfield Electronics

DOCTOR CLICK

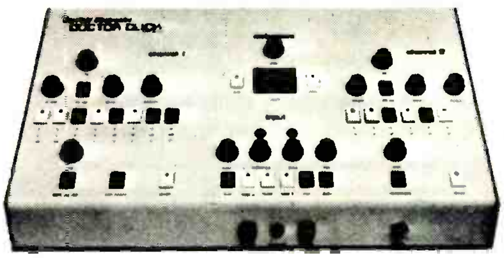
The **Doctor Click** Rhythm Controller makes it possible for the first time to synchronize the world of sequencer, drum machine, synthesizer composition with any one of the systems on the market or combinations of the systems on the market. Furthermore, the **Doctor Click** will cause sequencers, drum machines and synthesizers to play in time with a human drummer. It will also read click tracks and sync codes. The internal metronome provides both beats per minute and frames per beat calibrations.

THE DOCTOR CLICK RHYTHM CONTROLLER BREAKS THE BRAND BARRIER

SEQUENCERS		DRUM MACHINES		SYNTHESIZERS*		
DSX	Bass Line TB303	Linn LM-1	CR5000	Prophet 5	Modular Moog	Juno 6
Prophet 10	CSQ600	LinnDrum	CR8000	Prophet 10	OBX	Juno 60
Polysequencer	SH101	DMX	CR68	Prophet 600	OBXa	Polysix
Pro One	Emulator	Drumulator	CR78	Prophet T8	OB8	Poly 61
Model 800	Fairlight	TR808	KPR-77	Minimoog	JP4	Voyetra-8
Microcomposer MC4	Synclavier	Drumatics TR606		Memorymoog	JP8	
Chroma						

*(VCA, VCF, VCO, Gate, Trigger or Arpeggiator as provided on each unit.)

Measures
17½" x 11" x 4½" x 2½".
Weight is 8 pounds.



Warranty is one year.

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- 2 Metronomes
- 2 FSK Sync Code Decoders (Covers Linn, Oberheim, Roland)
- 2 Rhythm Envelopes
- Pulse Counter
- Pulse Shaper
- Gate Output
- Headphone/Speaker Output
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The brand to brand problems of timebase, voltage level and polarity are solved by the **Doctor Click**'s diverse output capability.

The ability of the **Doctor Click** to connect to many units at once coupled with its footswitch control capability makes it ideal for multiple sequencer, drum machine, synthesizer live applications.

Since the **Doctor Click** metronome produces beats per minute and frames per beat calibrations it is always convenient to get just the tempo you need. It is even possible to get fractional tempos such as 118½ beats per minute.

The **Doctor Click's** two independent rhythm actuated envelopes allow VCF, VCA and VCO parameters of synthesizers to be modulated in 32 rhythm values ranging from four measure cycle to 64th note triplet with variable attack, decay, sustain and amount. This eliminates the problem of rhythmic drift when using a conventional LFO.

The ability of the **Doctor Click** to transform metronome click tracks into timebase clocks allows frames per beat music film work to be done with virtually any sequencer, drum machine or synthesizer.

The ability of the **Doctor Click** to read live tracks allows sequencers, drum machines and synthesizers to play in sync with the varying tempos of a human drummer or a built click track.

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The pulse shaper circuit turns a pulse from an instrument into a trigger waveform allowing synthesizers to sync to a drum fill.

The headphone output allows click tracks in multiples of the tempo to be generated and is capable of driving a speaker.

The pulse counter can be used to program sequencers in higher timebases, quickly combining greater rhythmic resolution with step programming accuracy.

The step programming switch can be used to step program sequencers that normally do not have this capabilitv.

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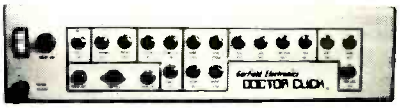

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PATENT PENDING

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"Night Line" . . . continued — Page #4

PRODUCTION SCORE SHEET FOR "SHADOW OF YOUR SMILE"

Arranger: Dick Hazard

strings are playing a high pad. The vocalist is still singing against the band, and there's a woodwind entrance at bar #15. This entrance tells me that the song is setting up for something. Again, we have slow harp glissando coming out of bar #16 and #17. Now we have flutes and vibes playing a figure together, which would tell me that the chart is starting to build a little bit, and I may have to write a note here to remind me to open up the rest of the woodwind mikes.

"We have vibes entering. I might have the percussion muted at that point, not so much for leakage, but for any kind of shuffling around that might be going on in that section; the guy bumping into his music stand, for example. Since the vibes enter at bar #17, I'll open up his mike probably about bar #16. The violins are now doubling up on a flute line, which means there's going to be more weight going down. At bar #24 the harp is not playing glissandos anymore — it's now playing lines — so he's digging into it a little more. I'll be watching levels because I suspect that things are going to get a little toastier.

"At bar #25 we're still building gradually. The violins are digging in a little bit more around bar #27, bar #28, and at bar #29 it's going into a Rubato [out of tempo] section, which means it's going to stretch a little bit. Here we have some dynamics; we have a crescendo into a decrescendo at bar #31 to bar #32, and triplets. Which means it's going to get real broad here.

"Ideally on the first run down, unless I didn't have any choice, I might have my hand on the faders. But I would try not to move anything, because I'd hope that the orchestra and the conductor would take care of the dynamics for me. I'm not looking to ride faders unless it's a *real* necessity, but I might mark on my chart that something is going to happen, and I have some dynamics marked.

"We have brass entering here for the first time at bar #33, and it looks like we've come down and are going back up again dynamically, with a big, full, round orchestra sound. Although there's no dynamics marked on the chart, we have more of a full orchestral sound starting to happen. Again I see the word triplet written on the chart [3], which leads me to believe it's getting broad again.

"On the end at bar #41 we have the trumpets in Harmon mute, which tells me the ending will be a softer one. Again there's no dynamics written here, but it's one of those tunes that has a lot of ups and downs, and is going to kind of end on that mellow note.

"As I say, during the first pass I would do nothing with the faders, although I'd try to watch the chart and make whatever marks I could as it went down. If I had to ride something on the next pass, I would know where. It's sometimes nice to have the musical advisor there to give

AURAL IMAGINATION

An engineer needs to acquire a sense of balance. Not only do instruments have different tone-colors, they also vary a great deal in terms of strength and power. The most powerful instrument in the orchestra is the trombone; the weakest the flute on its bottom notes. A note may be written and marked Fortissimo for trombone and the same for the flute, but the level difference would be overwhelming.

In modern scores, where niceties of balance are more carefully put down on paper than they were in classical times, and where infinitely more subtle nuances of tone are aimed at, it's common to find different degrees of loudness indicated for different instruments or groups of instruments, in order to obtain satisfactory balance and to ensure sufficient prominence for important lines in the texture, and sufficient effacement for the background. Doubling of several instruments in unison or octaves also is often used to ensure good balance, and to bring important melodies into sufficient prominence.

With regard to the ability to recognise the sound of each instrument, which of course is the first requisite of the scorereader, this can only be acquired by observation. Listening to an orchestra, score in hand, the reader will soon learn to recognise the instruments when he hears them, and thus will acquire the ability to call up to his mind's ear the sound of any simple instrumental combination. For example, the oboe is easy to recognize with its pungent "tang," but it needs some practice to infallibly distinguish flute from clarinet on certain notes, and softly played horns from bassoons. Wind instruments sound very different at different pitches, notably the clarinet, whose lowest register is hollow and sinister, and in complete contrast with its clear and pure tone in the middle and upper registers. The low notes of the flute also are very characteristic, and bear a marked resemblance to the tone of a quietly played trumpet.

Among the brass instruments, by reason of its slightly muffled and veiled quality, the horn is distinguishable from the trumpet and trombone, and even when played very loudly it never gives the open-throated and

PRODUCTION SCORE SHEET FOR "SHADOW OF YOUR SMILE"

. . . continued —

you cues, and other session directions. In the case of direct-to-disk albums it allows me not to have to watch the score, although I will read through it first. It lets me concentrate on the moves, and get a little more wrapped up in the music. I don't have to eyeball the chart so much if there is someone there to call the cues to me. In direct-to-disk it's necessary to know what's coming up because there's no preview signal for the cutting lathe, and the musical director is not only calling cues to me, but also to the lathe operators who have to open and close the groove width."

Full Scores with Conductor

Composers and orchestrators understand such abstract concepts as balance, weight and colors, and it is their job to balance the sound of the orchestra for an engineer. Rick Riccio engineers sessions for television and film music soundtracks at Evergreen Studios in Los Angeles. The bulk of Riccio's work is with large orchestras, although he does have to work with popular music as it relates to TV and film sessions. His musical instrument is flute, which gives him a big advantage in understanding the classical forms.

The ability to read a full score and understand it is important in his work, he says. "If you get the chance, try to run down the chart with the arranger or conductor. From the booth [control room] you have a golden opportunity to watch the conductor just like a player. You can take your dynamics from him, because he's going to be conducting the orchestra; he's also conducting *you* at the same time, and an engineer can tell a lot from his movements.

"Without looking at the chart, you should be able to watch the conductor, and have a real good idea what the dynamics are going to be. Also, which sections are going to be playing, because

. . . continued overleaf —

the conductor is going to turn to different parts of the orchestra. If he turns to the string section to the right, for example, there's going to be low strings there, and to the left the high-strings."

The conductor's purpose is to hold the musicians together throughout the entire piece, which encompasses meter, dynamics, shades of expression, style, and orchestral execution. Above all, his function is to see that the composer's intentions are adequately represented. The hands of a conductor are used in the following manner: the right hand keeps time, while the left is used to indicate entries, dynamics, and general expressions. These actions basically encompass two types of movements: the staccato or tight movement; and the legato, or loose movement. A baton generally is used when conducting a large ensemble as an extension of the right hand, so that the meter-keeping hand is very easily seen. The conductor's eyes and body attitudes also may be used to communicate his intentions to the musicians.

The accompanying chart for "Glad to be Unhappy" is an orchestrator's score, or sketch. In this initial form the various parts can be played on a piano. Later, individual detailed charts will be drawn up for each instrument or orchestra section. An engineer like Rick Riccio will work carefully with the orchestrator, making sure that all the musical sonorities are represented on the tape.

"Right up front," Riccio says, "the chart is telling me everything about orchestration: five alto flutes, two horns, eight brass, 12 violins, four violos, and four cellos. We're dealing with a pretty big section here. Because there is no rhythm, I would ask the arranger if we're dealing with a pure vocal/orchestra session, or is it an overdub situation.

"The chart starts off with Barbra Streisand singing; no intro. It has five alto flutes and two horns playing in a counter-melody, and it's probably going to be pretty soft. We have brass playing pads; violin and cellos playing high pads. Again there are no dynamics marked, so I would listen to the first

AURAL IMAGINATION

. . . continued —

impressively "brassy" tone of the latter instruments. Trumpets and trombones have an almost identical timbre, the trumpet tone being an extension upwards of that of the trombone, or vice versa. The tuba differs entirely in quality of tone from the trombones, though it is so often employed in combination with these instruments, usually acting as a double bass to the brass group; ie, doubling the bass trombone, which has the bass of the harmony, an octave below.

It may be helpful in learning the sounds of the instruments to provide a few examples of solo passages in well-known works, so that the reader may look out for them. Here are some, taken at random:

Flute: Debussy, "L'apres Midi d'un Faune"; opening.

Oboe: Mahler, "Song of the Earth"; opening of second movement.

Cor Anglais: Delius, "Dance Rhapsody Number 1"; opening.

Clarinet: Sibelius, "Symphony Number 1"; opening.

Bass Clarinet: Wagner, "Tristan and Isolde —King Mark's Song"; in second act.

Bassoon: Tschaikowsky, "Symphony Number 6"; opening. Stravinsky, "Le Sacre du Printemps"; opening (extreme high notes).

Double Bassoon: solos for this instrument are very rare, but a well-known example exists in "Beauty and the Beast," from Ravel's *Mother Goose Suite*.

Horn: Vaughan Williams, "Pastoral Symphony"; opening and letter L. Many examples of passages for four horns in unison are to be found in the works of Strauss, Mahler and Elgar.

Muted Horn: Stravinsky, "Petrouchka"; closing scene 4. Numerous examples will be found in modern works by Vaughan Williams, Bax, Walton, Milhaud, Honneger, and Prokofieff.

Trumpet: Benjamin Britten, "Sinfonia da Requiem" second movement.

Muted Trumpet: Moussorgsky; Ravel, "Pictures from an Exhibition No. 6."

Trombone: Holst, "The Perfect Fool"; opening of ballet music.

Muted Trombone: Strauss, "Salome"; closing scene.

Tuba: Wagner, "Faust Overture"; opening.

■■■

Full Score for "Glad To Be Unhappy"

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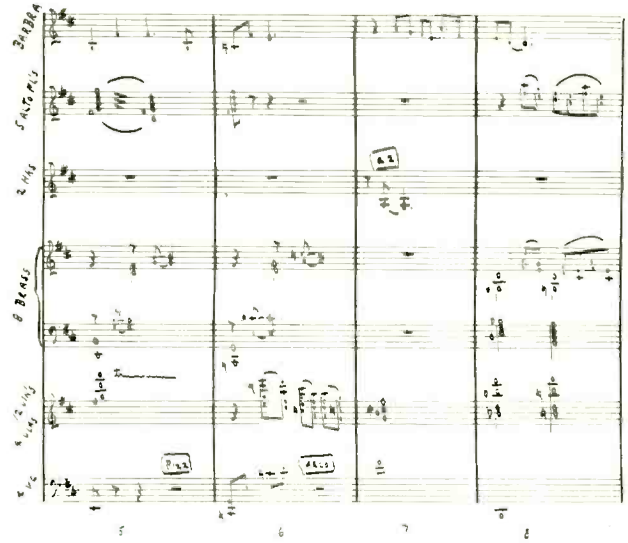
Arranger: Albert Harris

1



1-B

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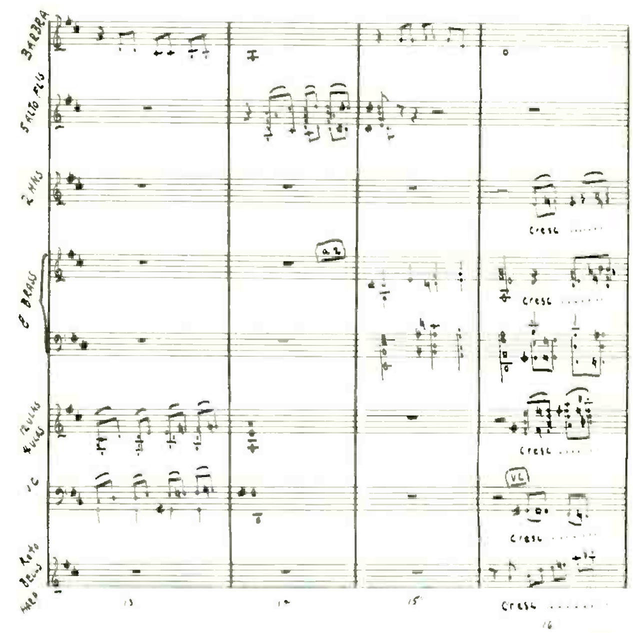

2-B

rundown and mark the parts.

"Starting at bar #5, I've got two pyramid chords: one on #5 and the other on #6. I have cellos going to Arco [bowing]. Again, it looks like it starts to open up as it gets into bar #17. We've got the brass doing punchs, and Barbra answering them, so we know that it's starting to get a little bigger. We've got the violins, violas, and cellos getting into a lot of movement, so again I can judge the chart's starting to stretch here. The violas and cellos are starting to play eighth notes, and the violins are written in pretty big block chords. I see movement; I see everybody going to eighth notes here, except for the flutes and trumpets and the brass, which are laying out.

"After bar #16 I see in the percussion that it's now marked koto, bells, and harp, which would lead me to believe it's going to start to sound a little exotic. From the last page it looks like it's not going to be the big ballbuster ending; we're dropping down to something real pretty — it looks like Barbra, strings, and koto. I would just give myself enough room going into this to let it grow, and come back down. I maybe would make a notation that I might have to ride the koto, which is a real soft Japanese string instrument. And be aware of the balance with the strings if it was a live session, although it wouldn't be that much of a problem with multitrack."

TV Cue Sheet

To examine the way in which Riccio would respond to music he hadn't yet heard, we asked him to look over the accompanying part of a television score. The chart is marked with the dynamics, and shows trombones, trumpets, horns, and six clarinets.

"The first thing I'd ask," Riccio says, "is how are you dividing up six clarinets on those two notes [laughs]. I'd say that it would way overbalance what is going on. Everybody is in forte, so it would *ah-three, ah-three* [half of the six clarinets], and that's a good balance.

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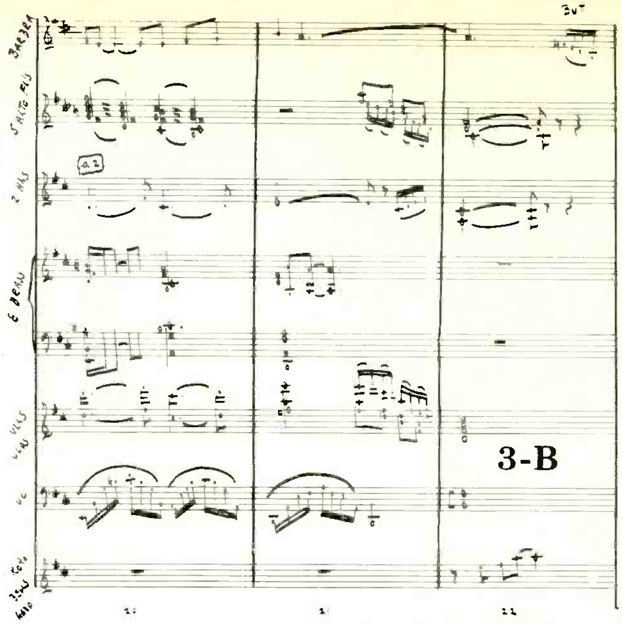
3



4

"What bothers me is when you're doing a feature, and the composer comes in and says, 'Okay, lets start with the title.' I *hate* to start with the main title, because the musicians aren't warmed up: nobody is set in the natural ambience of the room, and musically the whole thing doesn't happen. After a half-hour or an hour into the session they can get a better main title, since everybody is used to playing with one other, and are getting better balances. And it has a natural feel. When they start right off with the main title — which is the most important thing on the date — you don't have anybody playing as an ensemble; they've just come out of the cold, with downbeat at eight in the morning.

"I would sugget starting with some easy, unimportant cue with the full orchestra, so they can start to feel together, because the balances change dramatically within the first hour. Even if I don't touch a knob, acoustically the orchestra changes drastically. As they get warmed up to the room, the musical subtlties start coming out.

"The third bar of this music has a MP [messo piano], and the violins show. I don't usually ride the faders — I set my masters, peaks, fortes as much as I can. Then, when the pianissimo is live, I don't try to help them unless I really think there's a problem with noise. I'd raise the top-end slightly, but that's about it. I try to let it happen naturally unless it's not happening, then I would start easing in with the faders.

"There's this timpany part coming in later, so I'd look for that because it's right on the open alone. I'd also watch these string terraces coming in, and make sure everybody's matching. And also make sure that the forte piano attack spoke individually, and telescopes into the pad underneath.

"First of all I'd listen to the run through, and wouldn't even look at the chart. Then, if something was missing I'd grab the score, because an engineer rarely sits with the score unless he specifically asked for one.

"Really, I'm too busy with the floor guy [second engineer], who is running around asking the percussionist what he's going to be playing, and adjusting the mikes. During all that I'm presetting the levels, because if I know it's going to be a triangle part, and the percussionist will be using the same mike, we're going to be moving the mikes and establishing different balances. The two instruments won't sound good on the same mike, *unless* your getting the perspective right.

"Most of the percussion players are pretty good. If there's a problem with the mikes they will divide the parts up properly, and will acoustically balance themselves.

* * *

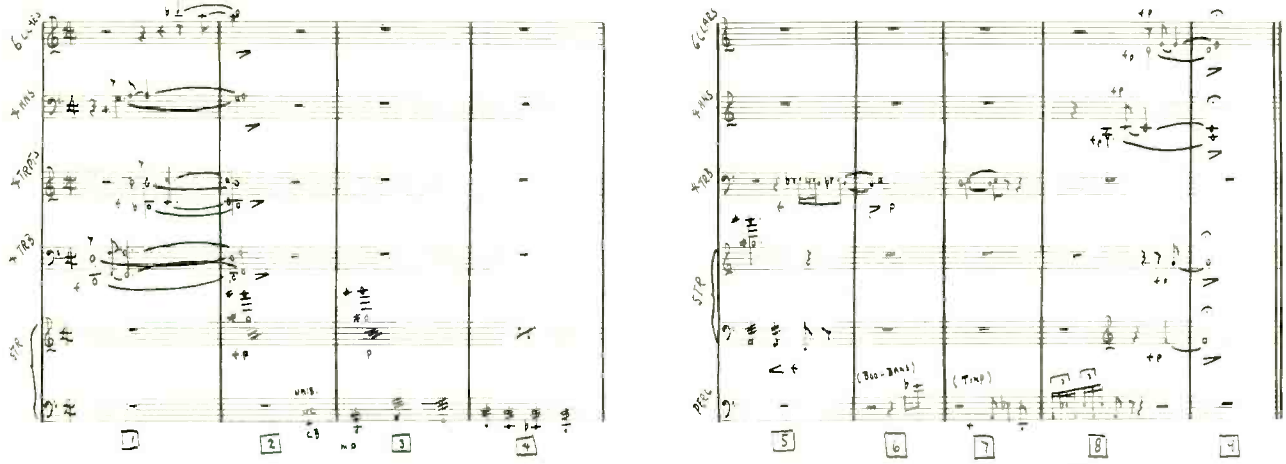
The need for an engineer to understand music during a session is best summed up by Bruce Swedien: "I will *always* sacrifice a technical value for a musical production value. *Music is number one.* In the relationship between musicians and producers, I think it's made it easy for me to work by keeping the musical values uppermost."

Recommended Further Reading

1. "Learn to Read Music," by Howard Shanet; Simon and Schuster. For the layman this book is excellent, and covers the skills needed for reading and understanding music essentials.

TELEVISION CUE

Arranger: Jimmy Stewart



2. "What to Listen for in Music," by Aaron Copland; Mentor Books. For the layman, Copland gives a full and clear explanation of the principle forms used in classical music. The chapter on tone color should provide a fuller understanding and appreciation of the various aspects of timbre created by musical instruments.
3. "Listening to Jazz," by Jerry Coker; Spectrum Books Prentice-Hall, Inc. The main thrust of this book is to help the reader understand the objectives and accomplishments of the best jazz improvisers, with a minimum of technical language.
4. "Sounds and Scores," by Henry Mancini; Northridge Music. This book contains printed scores of Mancini's recordings, which allow the reader to compare sound to the printed music illustrations. All the music illustrations are written in the "concert"; the same key as the piano.
5. "The Contemporary Arranger," by Don Sebesky; Alfred Publishing Co. By illustrating in scored and recorded examples, this book shows actual arrangements written for Dionne Warwick, Roberta Flack, and many other recording acts. The text explains problems in orchestrating for the record medium.

Recommended Recordings

1. *Tane Kain*; RCA AFL1-4381-A. Listen to the track, "My Time to Fly," and relate it to the chart provided in the article, listening for the arrangement and song form.
2. *Donna Summer*; Geffin Records. Listen to "Livin' in America"; read and follow Bruce Swedien's chart, noting his markings as they relate to the production of recording.
3. *Barbra Streisand and Other Musical Instruments*; Columbia PC32655. Listen to the "Glad to be Unhappy" track. The music that appears in this article starts after the free or Rubato section of the song. Listen for orchestral colors and balance.
4. *Seraphim Guide to the Instruments of the Orchestra*; Seraphim S-60234. This recording gives an idea of the basic tone qualities and range of each instrument of the symphony orchestra. It also demonstrates each instrument in isolation, providing the opportunity to hear it out of its usual orchestral context.
5. *Leonard Bernstein New York Philharmonic: The Young Person's Guide to the Orchestra*; Columbia MS6368. This recording introduces the instruments of the orchestra as they relate in family groups. The score is also available from publishers Boosey & Hawkes. When you listen and follow with the score, the experience will give you a fuller understanding of the various aspects of tone colors produced by the symphonic orchestra. ■■■

The way in which audio advances can be very perplexing. Often we can measure improvements in sound that we cannot hear, and can hear improvements that we cannot measure on our test equipment. By way of an example, the reduction of amplifier distortion from 0.08% to 0.05%, while measurable, cannot be heard, except by those rare engineers and producers who claim to have "golden ears." The difference in sound between two high-quality amplifiers is difficult, and sometimes impossible, to detect. It is not that sonic improvements are impossible, but rather that improvements in *measurable* parameters often seem to make little, if any, difference to the way the device sounds. There is, in fact, general agreement that the major coloration of sound comes *not* from electronics, but rather from the transducers at each end of the audio chain — the monitor loudspeakers, and the microphones.

Probably, because the need for improvement in transducers is more obvious, such devices seem to attract more attention and more unusual design approaches than the rest of the audio chain. Each design has one or more evangelists proclaiming the true path to "sonic salvation" to a fervent following of disciples. In the field of loudspeaker tweeter design, for example, there are proponents of horns, piezo tweeters, soft- or hard-dome tweeters, ribbon tweeters, electrostatic tweeters, and on and on. Each type of tweeter has its advantages and disadvantages, and may be brilliantly or poorly executed in its design and construction.

REALISTIC STEREO MIKING FOR CLASSICAL RECORDING

by Michael E. Lamm and John C. Lehmann

L² MicArray for PZM coincident-pair recording

The same may be said of microphones and miking techniques. There are two basic approaches to miking technique: multimiking, and minimal miking[1*2]. In actuality, each of these two basic methods is ideal for its originally intended purpose. Multimiking, since it provides greater acoustic separation between instruments, enables great flexibility in creating unique sounds through the use of overdubs, outboard special effects, and the layering or stacking of individually recorded tracks. In essence, with multimiking techniques the equipment and the recording engineer become *part* of the creative process; most modern recording sessions are done in this way. And because it is human nature to use all of the tools (or production "toys") available to us, and because we, as audio engineers, are creatures of habit, such techniques also have come to be used in the recording of classical music.

The science of psycho-acoustics is still in its infancy, and there is much to be learned about the ways in which we hear sounds. Some points can be stated with certainty at this time, however. We determine the source of a sound in the left-right horizontal plane by means of the brain's use of amplitude and time differences. Sounds originating from the left-hand side of a soundfield arrive at the right ear slightly lower in amplitude and later in time than those detected at our left ear, while sounds originating further away are delayed in time and lower in amplitude than close sounds. In addition, sounds produced at greater distance within an enclosed space also will have a higher ratio of reverberation to direct sound than closely originating sounds.

Only in the last few decades has music been written with the capabilities of the electronic media in mind; all music composed before the Twentieth century was written with natural acoustical performance in mind. And, traditionally, orchestras have been arranged with the louder and more directional instruments to the rear of the stage, the conscious or unconscious goal being to help the orchestra become self-balancing amongst the composite instruments.

This article will attempt to provide the reader with a broad overview of how we hear, provide a complete listing of the traditional microphone techniques that

— the Authors —

In 1978 **Michael E. Lamm** founded Dove & Note Recording Company, Houston, Texas, for the sole purpose of doing location recording of classical music, both locally and in Europe. **John C. Lehmann** has a Bachelor's Degree in Music (Performance), and over 15 years of experience in audio recording, as well as video production and broadcasting. As tonmeister for Dove & Note, he has been instrumental in the design and development of the L² MicArray described in this article.

The Celebrity™ Series

Microphones for every application.

There are so many brand names and types of microphones on the market, that it is often difficult to choose the best mic for your particular application..... Peavey solves this dilemma with the introduction of our new Celebrity™ Series Microphones.

Each microphone in the Celebrity™ Series — CD-20™, CD-30™ and HD-40™ offers special response characteristics for various applications. Each mic is also offered with a choice of four cables. Example, if you choose the CD-20™ for vocal application, you have the option of a 25 foot XLR low impedance cable WITH OR WITHOUT an integral on/off switch (CD-20LS™/CD-20L™) or a 25 foot high impedance cable (XLR to 1/4" phone plug) WITH OR WITHOUT the integral on/off switch (CD-20HS™/CD-20H™).

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We invite you to visit your nearest authorized Peavey dealer to take a look (and listen) or write Peavey Electronics Corporation, 711 A Street, Meridian, MS 39301 for a complete Celebrity™ Series brochure.

Peavey®

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have been developed to create a realistic stereo image, and introduce a new technique (or set of techniques) for the pursuit of that goal.

Minimal versus Multimiking

Far-field miking allows the natural sound of the acoustic environment to become part of the total recording. Due to their greater distance from the microphone array, those instruments toward the rear of the ensemble will have a more reverberant sound, thereby adding a natural sense of depth to the recording. In comparison, a multimike recording will produce an image of a two-dimensional wall of sound; in other words, a flat image. Adding artificial reverberation will help to recreate a sense of depth, but by only moving the wall of sound into the distance, or moving by it aurally into a larger room.

Before these authors are drawn and quartered by thundering hordes of enraged producers and engineers, in the interest of objectivity (and safety — ours!) we feel compelled to mention the positive aspects of multimiking. Of course, there are the obvious creative possibilities offered by multimike techniques. One can virtually re-orchestrate a symphony, given sufficient time, equipment, and budget. More modest effects include strengthening the string sections, and warming or tightening the coloration of individual instruments or sections. Multimike and multitrack techniques can produce a synthetic reality which, with care and attention, is quite pleasing; Bach played on a synthesizer and the scat singing of Mozart are just two examples that readily come to mind. But vinyl and plastic laminates can only imitate leather and wood, and "synthesized reality" is only an imitation of a musical performance.

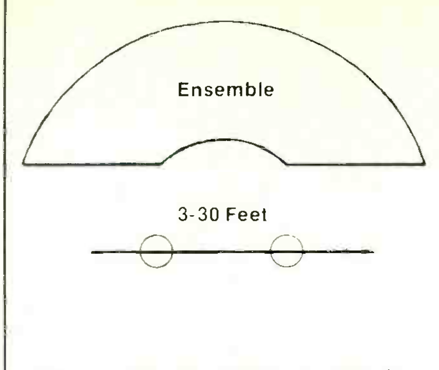


Figure 1A: Omni Spaced Pair

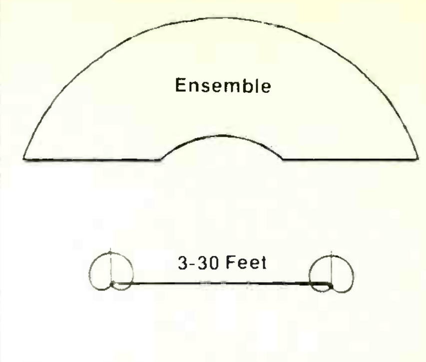
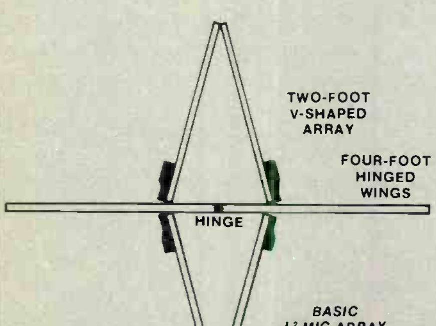


Figure 1B: Cardioid Spaced Pair

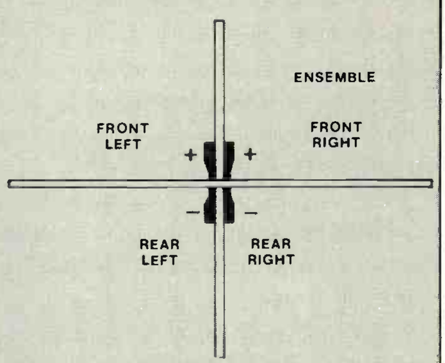
THE L² MICARRAY COINCIDENT PRESSURE-ZONE MIKING SYSTEM WITH VARIABLE PATTERNS

The L² MicArray consists of three basic segments: a pair of front and rear V-shaped arrays that together form a diamond, and which is divided by the third segment, that takes the form of a pair of hinged "wings." By rotating the wings forwards and backwards, and by opening and closing the front and rear V-shaped elements, an almost infinite number of coincident and near-coincident polar patterns can be set up.

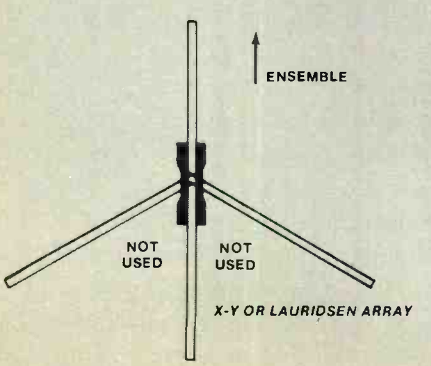
Shown here are eight possible arrangements of the L² MicArray, and which correspond to the "traditional" techniques described in the accompanying article: Blumlein, Lauridsen or X-Y, M-S, Faulkner-Blumlein, ORTF, DIN, NOS, and Binaural or Dummy Head. The polar response essentially is even from plate to plate, and virtually negligible in the area behind the plate. Bearing this in mind, it can be seen that some emulations are closer to the original technique geometry than others. The Faulkner-Blumlein emulation, for example, probably varies most from the original. It should also be kept in mind that there is a wide



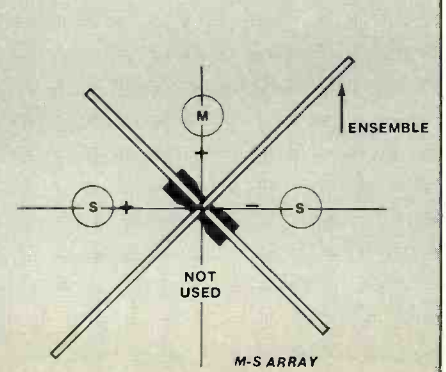
BASIC L² MIC ARRAY WITH FOUR PZM MICROPHONES



BLUMLEIN ARRAY
(NOTE: THE REAR PZMS ARE WIRED WITH REVERSE POLARITY. FRONT LEFT/REAR RIGHT FORM ONE FIGURE-EIGHT PICKUP PATTERN.)



X-Y OR LAURIDSEN ARRAY



M-S ARRAY

Therein lies its limitation, since there is a tendency in multimiked recordings for all the instruments to sound like they are only 10 feet away from the listener. Yet the mind knows that a 90-piece orchestra cannot sit within 10 feet of an audience. Also, there is a tendency for the sound imagery to be blurred — that is, for the location of the various instruments to be indistinct. (In a recent session we witnessed, which involved a combination of multimike techniques with far-field miking, instruments 30 feet away were made to sound 10 feet away on replay, while those 10 feet away ended up sounding as if they were 30 feet away.)

Perhaps because of the demise of quadrophonic sound, much thought has been given to the question, "What is stereo?" Or, more to the point, "What should stereo attempt to do?" Let's consider two possibilities. If we define stereo as internally related music produced simultaneously from two sources, then multimiking is well equipped to fill the bill. Many monophonic channels — single microphones — mixed down to two channels fits that definition nicely, and has given rise to the epithet "distributed mono." By the use of panpots, various channels can be distributed between the two available stereo channels. Such a distribution is on the basis of amplitude only, however, and contains no time clues for the brain to use in localizing the sound.

If we give stereo a much narrower definition, it might be phrased as follows: "Stereo is the realistic preservation and re-creation of the natural sound of a performance." This definition of stereo calls for the recording engineer to be both a scribe and an archivist, since an engineer's goal within such a context is to preserve the sonic qualities of an event as accurately as possible for eventual retrieval and replay at a later time. As such, the engineer must scrupulously *avoid* any influence on the signal, a role which is diametrically opposed to the actively creative role of the multitrack/multimike session engineer. We believe that it is extremely difficult for an engineer to switch back and forth between an intensely creative role, and an extremely passive one. There is always the temptation to "correct" the balance, boost the bass, or otherwise "sweeten" the sound to make the final product "better" (by someone's subjective criteria) but less accurate than the original. As a result, two very separate camps of devotees have sprung up: one seeing itself as *pragmatic*, making use of the available tools to produce a commercially viable product; and the other viewing itself as the *purist*, trying to document the efforts of artists who have spent years achieving the level of artistic interpretation at which they perform.

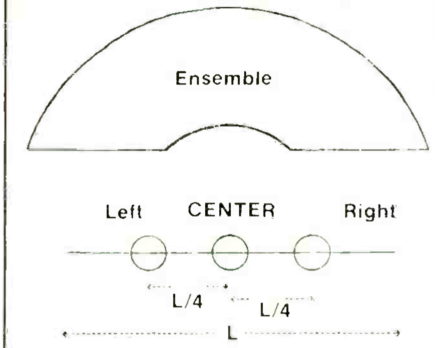


Figure 2: C.R. Fine

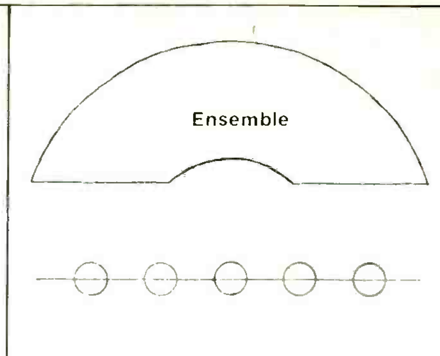


Figure 3: "Wall of Sound"

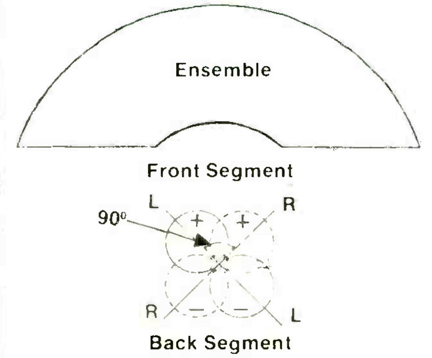


Figure 4: Blumlein

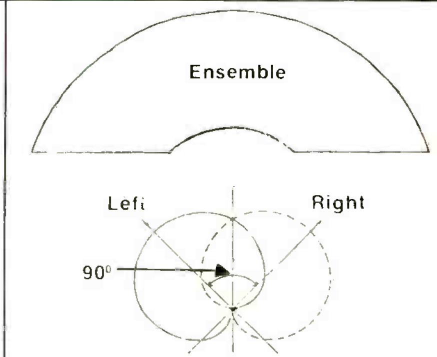


Figure 5: Lauridsen, or X-Y

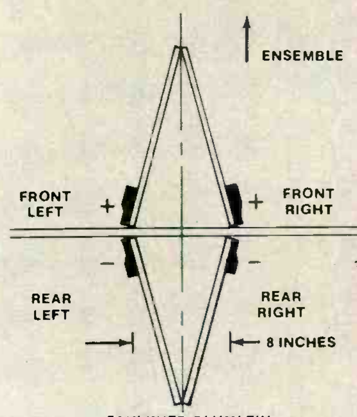
Realistic Stereophony

For over 50 years, the achievement of natural-sounding stereo reproduction has been the quest of many; an elusive goal that gets closer and closer, but which remains slightly out of reach. Lynn T. Olson possibly put it best when he said: "Accurate stereo records are still few and far between — probably less that 0.1% of those available . . . Great performances of present-day artists are being recorded with techniques that are directly comparable to hand-colored photographs — appealing and enjoyable, certainly, but in no way

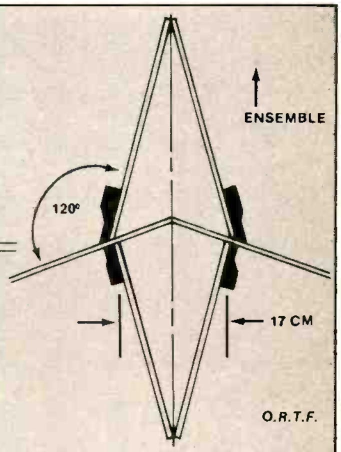
THE L² Mic Array — *continued . . .*

variation in actual polar response between standard microphones. Cardioid is a general term, for instance, and not an exact description of the polar response at all frequencies. In the same manner, omnidirectional microphones usually are quite directional at some frequencies, and vary greatly from manufacturer to manufacturer, and from model to model, in the degree with which they approach omnidirectionality.

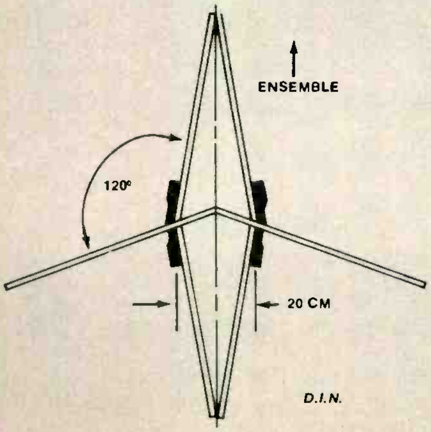
At this point in the development of the L² MicArray, some benefits are becoming apparent. A sound source is either on-axis to a PZM capsule, and is picked up with a virtually flat frequency response, or it is off-axis (behind the plate) and virtually unheard. The off-axis coloration exhibited by standard microphones used in conventional techniques is virtually eliminated. The other benefit is that during a live recording, audience noise can be reduced drastically by positioning the array so that the audience is located behind the plates. ■■■



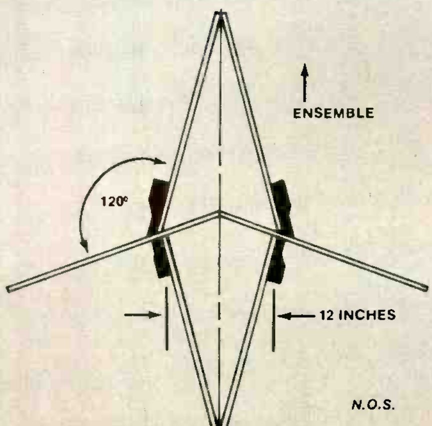
FAULKNER-BLUMLEIN
(NOTE: FRONT-LEFT/REAR-LEFT FORM ONE FIGURE-EIGHT PICKUP PATTERN.)



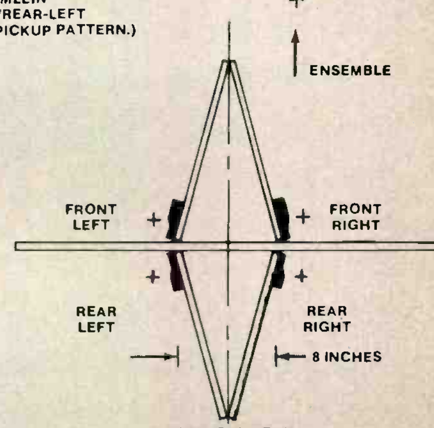
O.R.T.F.



D.I.N.



N.O.S.



BINAURAL HEAD
(NOTE: FRONT-LEFT/REAR-LEFT ARE ADDED AT EQUAL LEVELS TO BE THE LEFT-HAND CHANNELS; SIMILARLY FOR THE RIGHT-HAND ONE.)

accurate[3]."

Many "natural stereo" techniques have been devised over the years. Some are more successful than others at achieving the desired result[4], and most have their ardent supporters. All of the microphone techniques described in this article are useful tools, and have been utilized with varying degrees of success depending upon the performance circumstances, the equipment, and upon subjective criteria imposed by the listener[5].

In general, these natural stereo techniques are best suited for recording classical music, such as symphonic and chamber music, as well as choral works, pipe organs, and concert band music. In broad terms, however, any musical performance in which the ensemble to be recorded is responsible for achieving a homogeneous balance of sound would serve as a prime candidate for natural stereo techniques. Even jazz groups that have performed together for some period of time, and have developed their unique blend and balance, can be successfully recorded in this manner.

Microphone placement used for the techniques to be described is in the "far field" rather than in close proximity to the individual musical instruments, as is often the case in multimiking. Distant location of microphones avoids one of the major pitfalls in close miking —the tonal coloration resulting from being close to any specific area of an instrument [6] [7], and the inherent proximity effect of many microphones themselves, which causes a bass boost in the frequency spectrum.

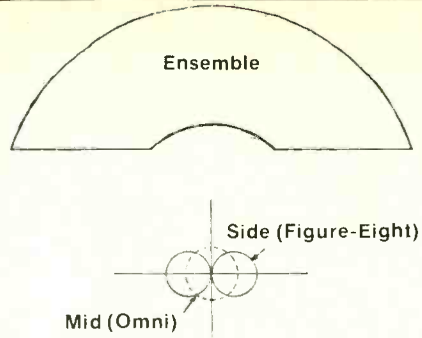


Figure 6: M-S

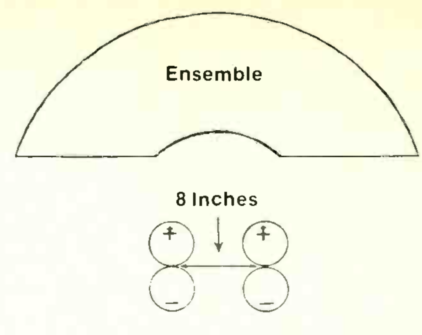


Figure 7: Faulkner-Blumlein

Far-field techniques can be divided into three main categories: widely spaced array; near-coincident arrays; and coincident microphone techniques.

The general criteria in selecting a microphone is flat on-axis frequency response, with smooth and consistent frequency roll off as off-axis orientation is approached. In addition, good transient response will help preserve clarity in the recording. From a practical standpoint, a microphone with a switchable pattern will allow the recording engineer to try most of the various techniques which to be described below. Microphones that feature switchable response patterns include: AKG C414,

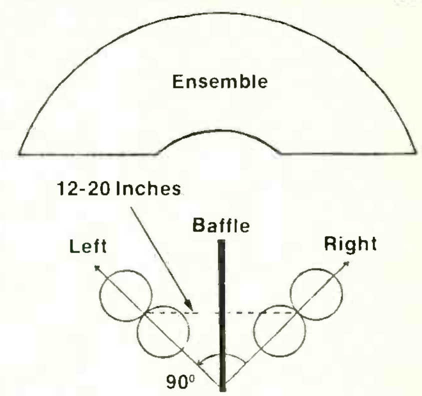


Figure 8: Madsen Shadow

MILAB DC-63, and Neumann KM-86, KM-88, U-87, and U-89. A less expensive alternative would be to use microphones with replaceable capsules; some examples include the AKG C450 Series, Beyer MCM Series, Nakamichi CM-300 and CM-1000 Series, and the Schoeps Colette Series.

WIDELY SPACED ARRAYS

1. The simplest arrangement is the spaced pair, shown in Figure 1, which consists of a matched pair of microphones mounted between 3 and 30 feet apart. A common practice is to space the microphones at a distance equal to one-third the width of the ensemble to be recorded. Russell Borud, a location recording engineer based in St. Paul, Minnesota, suggests spacing the microphones at least 12 feet apart. Spaced pairs that are too far apart can result in "ping-pong" stereo with a hole in the middle, while placing the mikes too close together can cause phase cancellations due to comb filtering, resulting in unnatural timbre. Experience with the technique will develop an intuitive sense of the best spacing. The choice of omnidirectional or cardioid polar response patterns for the pair of spaced mikes is dependent on the type of environment in which the recording is being made — for example, if there are sound reflections from rear and side walls that may find their way into the microphone, or if audience noise in back of the selected mike position is too distracting.

2. A classical variation on the spaced pair was pioneered by the late C.R. ("Bob") Fine (Figure 2), for which a center microphone is added to the spaced-

ADAPTABLE RIGGING SYSTEM FOR OVERHEAD COINCIDENT MICROPHONE ARRAYS

by Bert Spangler, Audio Coordinator, Media Development Center, University of Wisconsin-Eau Claire

There are about 75 performances held per year in the University of Wisconsin-Eau Claire's Gantner Concert Hall, each of which is recorded for subsequent critique by the performer. With such a performance schedule, the time and trouble associated with microphone placement is significant. Microphone stands certainly facilitate relatively easy placement, but have not been used very often because of their negative appearance to an audience; the risk to the safety and security of the microphones (and the safety of performers); and the difficulty sometimes encountered in finding a clear spot of floor on which to place them. Instead, we have chosen to hang the mikes from the ceiling or catwalks above the halls.

While this approach has eliminated all of our major problems, we still had to consider the time it took to hang the mikes. In response, a system of motorized hoists has been constructed that facilitate very quick microphone placement and positioning. With the use of a small hand-held radio control unit, a mike line can be "called down" from above, a microphone put on it, and the line raised to the elevation desired, all in a matter of a minute or two. If the microphones(s) need to be positioned upstage or down, stage left or right, an additional black nylon line or lines may be clipped on to the microphone cable before it is raised, and small winches on the other ends of the black nylon lines activated with the radio control to facilitate such positioning (see main diagram). In addition, the line connected to the coincident-pair microphone array can be rotated clockwise or counter-clockwise (as viewed from above), to provide the directional orientation.

Being unaware of any commercially available equipment for this purpose, the entire system has been fabricated by electronic technicians in our Media Development Center. Although a significant

Figure 1: Modified synchro motor mounted beneath roof, showing slip-ring connections to microphone cable (left).

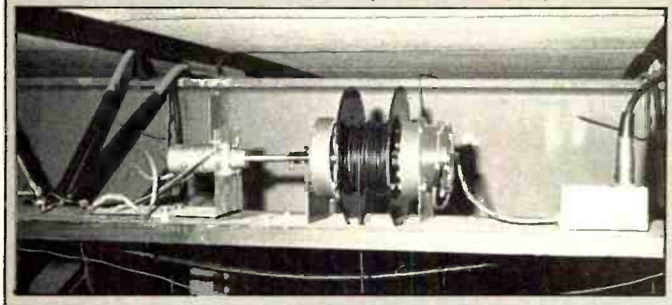
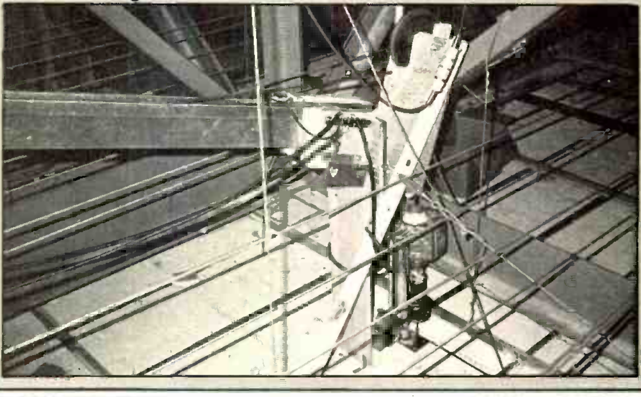


Figure 2: Custom-designed aluminum trough for dispensing and retracting 30 feet of 12-conductor mike cable.



For additional information circle #68

pair technique. This approach is suggested by, amongst others, Nakamichi in its advice to amateur recordists. Variations from this technique include departing from symmetry, and placing the three microphones at various heights or distances from the sound source. By way of an example, the fine recordings of the Eastman Wind Ensemble on Mercury Records were made by Bob Fine using three microphones. Adding a middle microphone

ADAPTABLE RIGGING SYSTEM FOR COINCIDENT MIKE ARRAYS — continued . .

amount of time has been invested in the construction and maintenance of this system, the materials cost was minimal. Some of the more costly items were obtained through a Federal Surplus Property Distribution Center.

Because of limitations of space, no attempt will be made here to document this system in sufficient detail to permit duplication of what we have done. Instead, it will be limited to a description of how we overcame the main construction obstacles, and anyone interested in developing similar equipment can build the power supply and controls they need.

Obstacle #1: What kind of electro-mechanical mechanism will enable, in a limited space, the dispensing and retracting of 30 feet of balanced microphone cable? The solution was found by modifying government-surplus selsyns, or synchro motors. The main reason for this choice was the motor's three slip rings fitted to the armature shaft, which provided the reliable sliding contacts needed to feed the microphone signal from what would become a small cable drum (Figure 1). The motor's field winding and its housing were removed and, after attachment of side flanges, the armature became the core of a small drum on which the cable is wound. This drum is then driven by a small reversible-gear motor coupled to the armature shaft. The integrity of the contact of brushes and slip rings has been greater than ever imagined. After many years of operation, there is still no noise generated at these contacts whether they are stationary or moving. (They are almost always used with phantom-powered condenser microphones.)

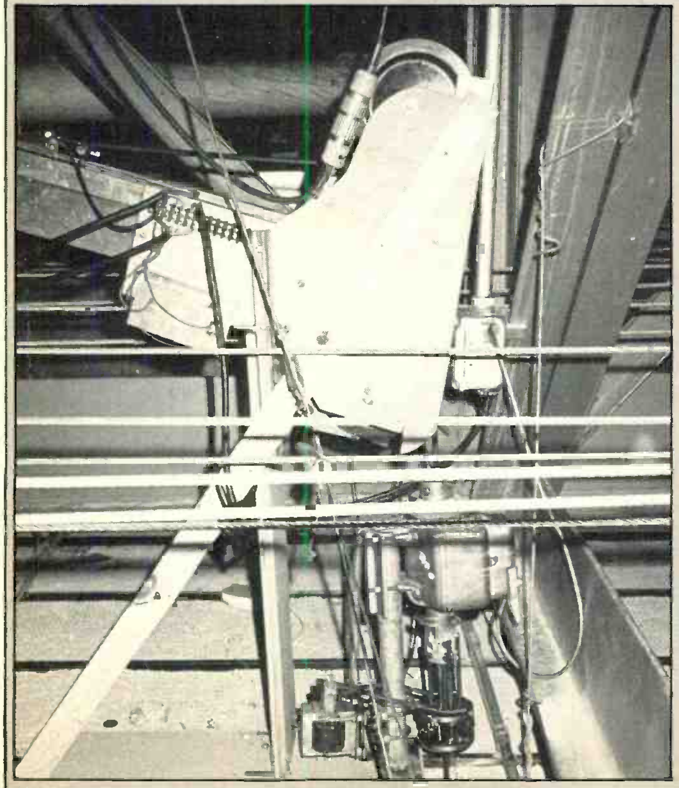
Obstacle #2: What kind of mechanism will enable, in a limited space, the dispensing and retracting of 30 feet of 12-conductor cable for a multiple-pattern coincident-pair microphone? The solution here was to construct an aluminum trough to serve as a track for a small cart on which was mounted an pulley, and the cart is let down or pulled up the trough by a small winch at the upper end of the trough, thus lowering or raising the cable (Figure 2).

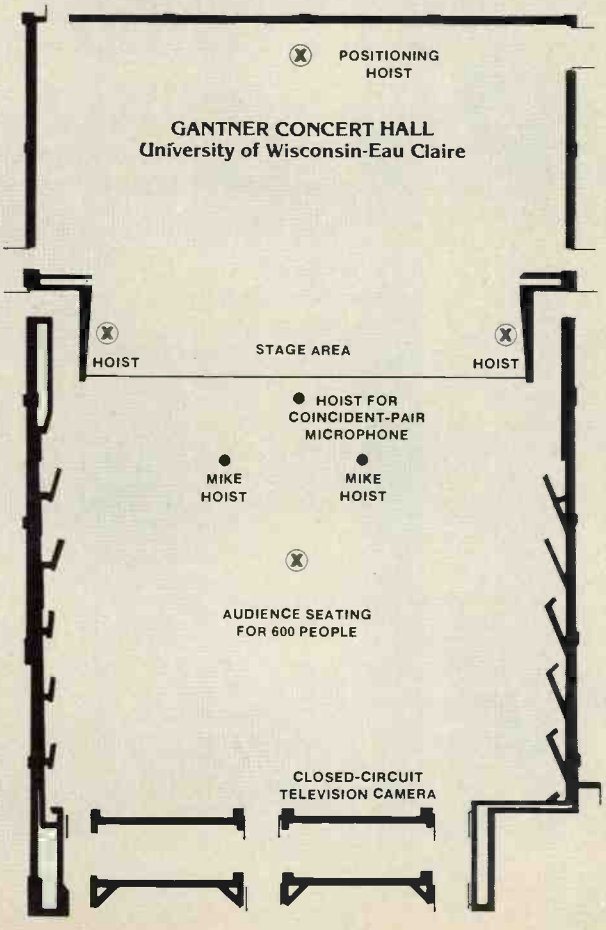
Obstacle #3: What kind of mechanism will make it possible to control the directional orientation of the 12-conductor cable, and not interfere with raising and lowering of the cable? A standard home televison antenna rotator was the answer here. The opening for the antenna support pipe was large enough in diameter to pass the unusually large connector of the microphone cable. The rotational speed was adequate; the reversing capability was there; and the price was right. However, neither of the first two obstacles was overcome without some custom modifications, and the same was true here. A clutching mechanism was constructed on the short rotator pipe, as detailed in Figure 3. When a raise or lower command is given, the solenoid on the clutch releases the cable so that it is free to slip through the pipe. Then, after a slight time delay, the cable begins to go up or down. Likewise, when the raise or lower command is ended, the cable stops moving and, after a time delay, the clutch solenoid is de-energized, the cable seized, and then responds to the turning of the antenna rotator.

I do not have space in this short article to describe further details of the rigging system, such as operation of the radio control unit, how so many control functions are handled on what is basically a four-channel unit, use of limit switches, and the manner in which the hoists operate at a fast speed for long distances, and a slower rate for final positioning. But these refinements are not essential to the operation of the system.

If you have a heavy schedule of recordings in a particular hall and would like to try this approach to microphone placement, my advice would be to start simple and get it functioning. Then, as time permits, add the extras. There are many innovative people who can improve upon the ideas presented here. Maybe you are one of them!

Figure 3: Side view of cable trough, rotator (below cables), and clutch mechanism (mounted just above ceiling exit hole).





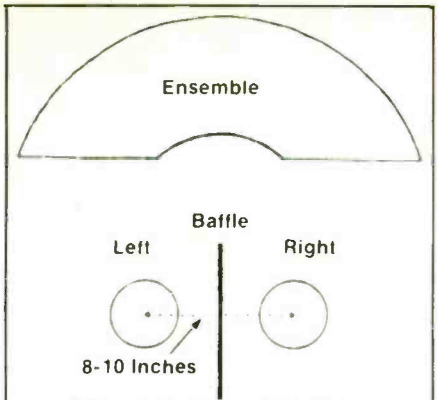


Figure 9: Borud Shadow

eliminates the ping-pong, hole-in-the-middle effect, but must be used judiciously to maintain realistic imagery.

3. A third approach in this classification is the "wall of sound," or "iron curtain" technique, shown in Figure 3, which places four or more microphones across the front of the ensemble. Although a far-field technique, it is debatable whether such an array truly can be called "minimal" miking. As more microphones are added phasing problems become greater, and also can cause difficulty in record mastering and FM broadcasting.

COINCIDENT MIKING TECHNIQUES

Coincident techniques use two (or more) microphone capsules placed in the closest possible physical proximity. Both microphones are mounted in the same vertical plane, equidistant from the source, with one directly above the other.

By virtue of the close capsule proximity, coincident techniques eliminate problems caused by phase cancellations — but at the expense of containing no time difference clues for the brain to use in localizing the source of a sound.

1. New ideas are not necessarily better, nor should our industry revere old concepts simply because of their age. Rather, new and old ideas alike should be constantly re-evaluated for validity. It is interesting to note that there is a resurgence of interest in the work of the British inventor Alan D. Blumlein, who patented a coincident technique in 1931. Figure 4 shows the layout of the Blumlein arrangement, which consists of two figure-of-eight microphones placed in a coincident array at 90° to one another, with the positive lobes angled 45° left and right.

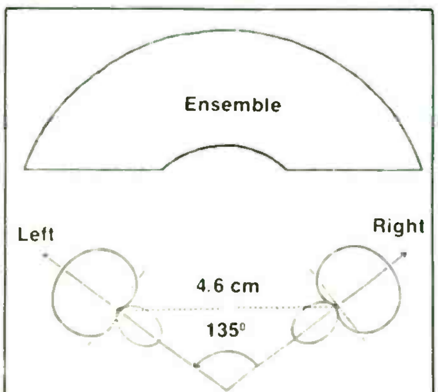


Figure 10: Stereo-180

2. The Lauridsen pair or X-Y pattern, shown in Figure 5, is another traditional approach that uses two cardioid microphones in coincidence, with their axes orientated 90° apart. Various other angles can also be used[4]. The X-Y technique generally results in a fairly even soundfield that is narrower than the spacing of the monitor speakers. As a general rule, the greater the angle between the mikes, the wider the resultant image. It should be noted that experimenting with various orientations of Blumlein (figure-of-eight) and X-Y (cardioid) coincident arrays can help control unwanted sound pickup in reverberant environments, and to reduce audience noise from the rear-facing figure-eight lobes, should this cause problems during a recording.

3. A third traditional approach is the M-S (mid-side) technique, shown in Figure 6, which uses a figure-eight mounted with the off-axis toward the ensemble, and an omni or cardioid placed coincidently. By using an electronic matrix device, such as the unit made by Audio Engineering Associates[8], or by matrixing the signals through a mixing console[9], one can electronically create Blumlein and X-Y patterns, as well as

ns in between. All of which he M-S particularly versatile for nt miking.

dent microphones containing ially rotatable) capsules include AKG C-422, C-34, and C-33, MILAB XY-82, Neumann SM-69, and USM-69, and Schoeps CMTS 301U, and CMTS 501U. Microphones designed for the M-S technique include the Fostex M-22, and Sony ECM-939, and ECM-989.

NEAR COINCIDENT ARRAYS

This family of techniques places the microphones at equal distances from the source, and at equal heights, but spaced apart horizontally at some small distance — usually 12 inches or less.

1. Faulkner-Blumlein (Figure 7) utilizes two forward facing figure-of-eight microphones spaced eight inches apart, and is a technique that works well at greater distances from the ensemble than most. The resulting image tends to be placed between the monitor speakers, with a slightly emphasized center.

2. Madsen Shadow (Figure 8) utilizes two figure-eight microphones separated by a baffle, and angled at 90⁰ to one other.

3. Borud Shadow (Figure 9) uses two omnidirectional microphones separated by a baffle covered with Sonex acoustic foam. A variation on the Madsen Shadow technique designed by Russell Borud, it tends to produce a fairly wide sound image.

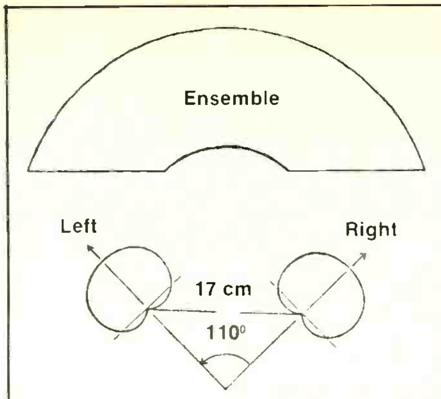


Figure 11: ORTF

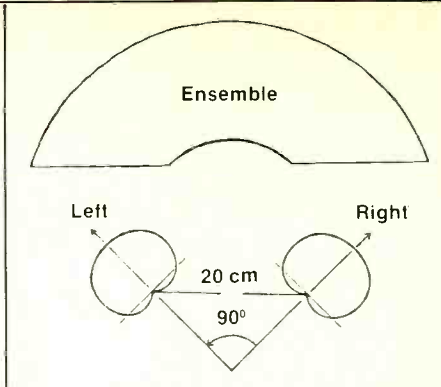


Figure 12: DIN

4. Olson Stereo 180 (Figure 10), involves the use of two hypercardioids angled at 135⁰, and spaced 4.6cm (1.8 inches) apart[3]. Since polar patterns vary for different hypercardioid microphones, it is worth experimenting with various inter-mike spacings, and orientation angles.

5. ORTF (Figure 11) utilizes two cardioids spaced 17cm (6.7 inches) apart, and angled at 110⁰. The technique was developed by and named after the French national broadcasting network. Some recording engineers have observed that the ORTF technique produces in a hole in the middle of the resultant soundfield. It has been our experience, however, that this effect is dependent upon the "fatness" of the

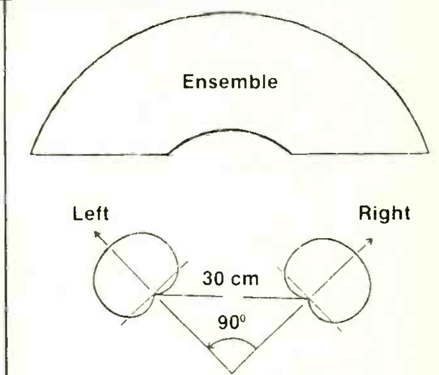


Figure 13: NOS

cardioid mikes being used (in other words, the width of their on-axis polar patterns at various frequencies), and can be reduced by selecting appropriate microphone models.

6. DIN (Figure 12) makes use of two cardioids spaced 20cm (7.8 inches) apart, and angled at a 90-degree angle[10].

7. NOS (Figure 13) utilizes two cardioids spaced 30cm (11.8 inches) apart, and angled at 90⁰. Developed by and named after the Dutch national broadcasting network, the NOS coincident technique can lead to phasing effects because of the wide capsule separation, if placed too close to the ensemble. (As a general rule of thumb, the distance between spaced microphones should be less than 12 inches, or greater than 12 feet.)

8. Binaural Head utilizes an artificial human head with microphones mounted in the ear openings, as shown in Figure 14.

Spacer bars for setting up microphones in techniques #1, and #4 thru #7 are available from Beyer, Shure, Schoeps, Atlas, and Radio Shack. Binaural or Dummy Head microphones include the Sennheiser MKE-2002, and Neumann KU81.

It should be noted that there is a great deal of similarity amongst many of the near-coincident patterns. Spacings vary generally from two to 12 inches, and the angle between the microphones from 90⁰ to 135⁰; also cardioids or hypercardioids often are used. There is a great deal of variation in polar response patterns from one model of cardioid to another, and the same is true of hyper-

Early prototype of L^2 MicArray utilizing large sheets of plexiglass.

cardioids. Therefore, the resulting imagery may depend as much on the polar response of the mike, as it does on the near-coincident geometry used. Only experimentation with a range of microphones will allow you to hone any particular technique for the best effect.

Some theoretical argument can be made espousing the near-coincident techniques, if we remember one rather obvious fact. All that we have ever heard, and all of the aural information which we perceive (pitch, distance, loudness, spatial location, etc.), is based on two receptors, one located on each side of our head. All imagery is based on time and amplitude clues which our brain decodes into spatial information. It can be argued, therefore, that the recording method which best preserves both the amplitude and time clues received by our ears is the one most likely to provide a realistic image. The near-coincident techniques are dependent upon both time and amplitude differences between channels, and thus theoretically would seem to hold great promise.

PZM Techniques

A word about PZM or pressure-zone microphones in general would be in order, since there are several models available which vary in their frequency response and application. As a result one should be careful about generalizations applied to a whole line of microphones, no matter who the manufacturer is.

Reviewers have given high praise to PZM microphones, remarking that "response varies very little as a function of either angle or distance from the sound source."[12] Phrases such as "excellent reach," "clear, sharp details," "crystalline," "no better microphone for chorus," and "crisp" also have been used. In the past year MILAB, Sennheiser, and Schoeps have all introduced microphones designed to take advantage of the boundary-layer principle.

As most engineers already are aware, a PZM is a small capsule mounted facing, and very close to, a flat boundary plate, and has a hemispherical polar pattern. The hemispherical pickup would seem to eliminate its use in many of the traditional techniques that called for specific polar patterns. However, the development of several prototypes by Ken Wahrenbrock, which used two and three intersecting planes or boundaries, showed that the directional pattern of the PZM could be shaped.

It should be stressed that in going from a single PZM mike — either mounted on a flat surface, such as a gobo floor, or a control-room window, or on a large backing plate — to a back-to-back stereo PZM pair (referred to by Crown as a Stereo Bipolar Plate) the resultant spherical pick-up pattern exhibits outstanding mono/stereo compatibility. Unlike the single-PZM array, for which there will be a 6 dB LF roll-off at the frequency corresponding to the dimensions of the backing surface, at low frequencies a stereo PZM array will exhibit a 6 dB recombination, due to the two overlapping hemispherical polar patterns combining at the plate boundary. Thus at low frequencies the stereo soundfield is purely coherent, and collapses perfectly to mono, thereby simplifying the disk-cutting process.

About two years ago, we began to

investigate the possibility of combining the assets of the PZM with some of the traditional microphone techniques. The goal was to make more realistic recordings, both from the standpoint of clarity of response, and clarity of stereo image. Several prototypes have been constructed to emulate traditional techniques, and the results have been most gratifying.

The ORTF technique is perhaps the most widely known near-coincident arrangement, and our initial goal was to adapt it for use with PZM models. The first prototype was built from an entire 4- by 8-foot sheet of ¼-inch acrylic plastic; it was four feet high, almost five feet wide, and with the necessary framework and fittings weighed 50 pounds (Figure 15). Although its size and weight made it difficult to transport to various recording venues, the array's sonic qualities were most encouraging. The first recording made with it was of a concert by the Texas Chamber Orchestra in a small opera house built in 1894. The resultant recording won an Honorable Mention in the classical division of the 1982 Crown PZM Challenge.

A second and third prototype followed in which we managed to reduce the weight to about 13 pounds. The whole array is constructed so it can be folded and packed into a case measuring four by 16 by 48 inches for carrying from one concert site to another. It has been used extensively, both mounted on a tall stand and flown from the ceiling, to record a wide variety of choirs and orchestras in many different types of locations. The results have been very good, in terms of depth and spatial separation of the soundfield, due in no small part, we feel, to the smooth frequency response of the PZM mikes.

Knowing that the various traditional techniques each have their own advantage, and that different situations call for different techniques, we were eager to try other PZM adaptations. In November of 1982, we were called in as microphone consultants for a recording of Handel's *Messiah*, which was broadcast nationwide by PBS on Christmas Day, 1982. The performance by the Concert Chorale of Houston, with the Texas Chamber Orchestra, was video taped and recorded on multitrack for subsequent mixing and post-production for the broadcast.

An NOS configuration was custom built for use as the main stereo pickup (Figure 16). Several design changes were incorporated for this specific project, in order to minimize visual obstruction by the mikes in the wide camera shots. The PZM capsule used also was upgraded to the Crown Model 6-S, because of its small physical size and

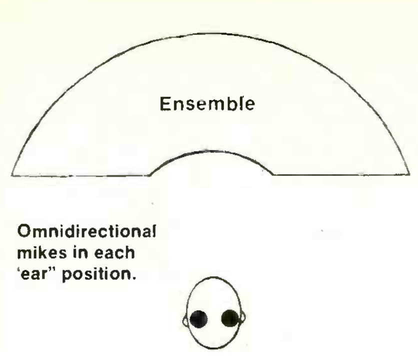


Figure 14: Binaural Head

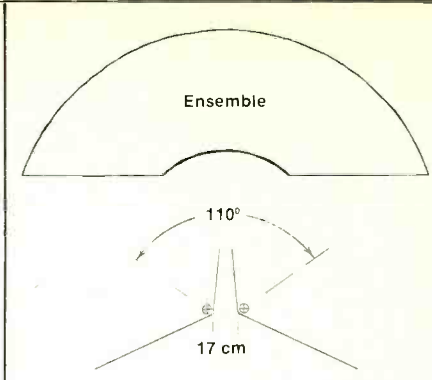


Figure 15: PZM-ORTF

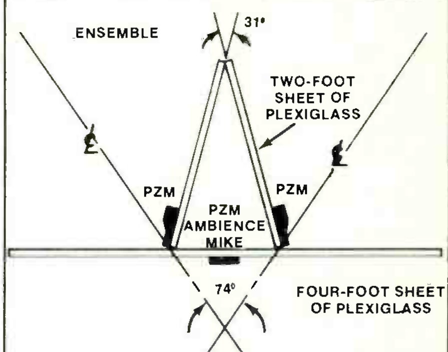
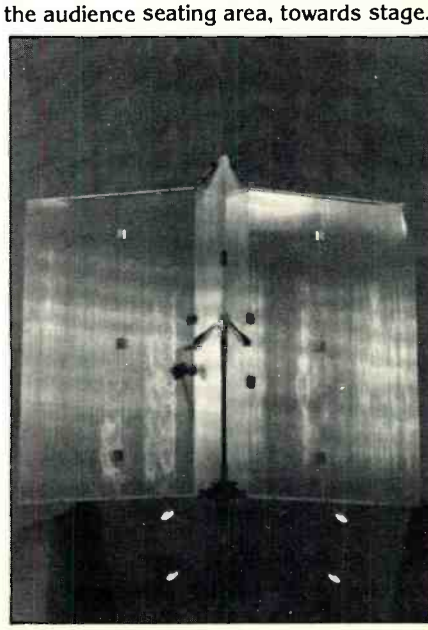
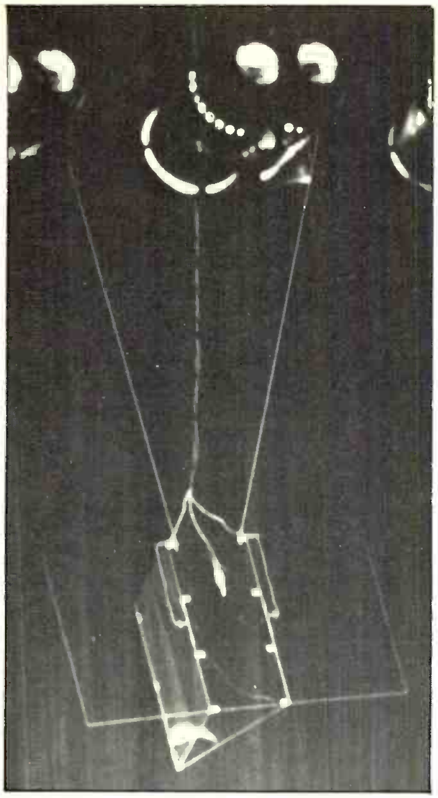


Figure 16: L² Mic Array using three PZMs, built for *Messiah* concert.

A later version of L² MicArray, flown above the audience seating area, towards stage.





L² MicArray used to record Handel's "Messiah" concert (detailed in Figure #16).

flat frequency response. During post-production mixing, many of the accent or fill mikes were not used at all, and in the final mix the NOS/PZM tracks furnished the bulk of the sound. The stereo image is broad and consistent across the space between the speakers, with good localization of sections and instruments. Combined with the clarity and accuracy of the newer model PZMs, the resulting quality and realism are delightful.

Currently under development is an array which allows the emulation of eight of the traditional patterns — Blumlein, Faulkner-Blumlein, X-Y, M-S, ORTF, NOS, DIN, and Binaural — as well as variations in between. This L² MicArray should prove to be a very versatile tool in the hands of the purist recording engineer, as well as a high quality main stereo pickup in multimiking situations. The development and application of this device is described in an accompanying sidebar.

The authors have a strong proprietary feeling about the L² MicArray, and anticipate applying for a patent once development is completed. Those who would like to participate in the field testing phase are most welcome to contact us concerning acquiring an L² MicArray for use on recordings. ■■■

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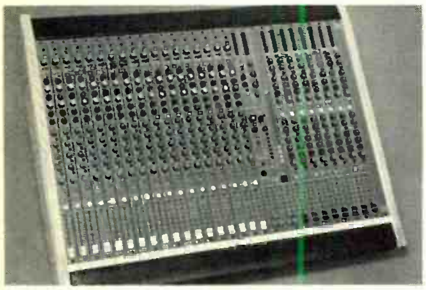
New Products

SOUNDTRACS 80 SERIES MODULAR MIXERS

Unlike conventional mixers that use mechanical switches to assign channels to either sub-groups or masters, each subgroup on the new 80 Series is a stereo pair, and each monitor section of the sub-group can be switched to a tape return — allowing, with the two auxiliary returns per sub-group, 24-track tape monitoring.

Although each channel has only *one* routing switch, by the inclusion of a microprocessor in the system an input channel can be routed to as many sub-groups as there are in the mixer — as well as the master. Additionally, the auxiliary returns are all assignable, via the microprocessor, to the sub-groups and masters.



To interrogate the system status there are four options:

- Press the routing switch on each channel or the aux return: the sub-group master it is routed to will display via an LED;
- Press the sub-group (pair) or master: the input channelss/auxiliary returns assigned to it will display via an LED;
- Via a UHF output the sub-group/input channel assignments can be displayed on a TV monitor;
- Via a video output the mixer can "handshake" with a video display complete with keyboard; by inputing the function of each input channel (1 = keyboard, 2 = tom, etc,), the display will write the sub-group format.

Since the microprocessor is programmed on an 8-bit bus to "seek" 36 input channels and 12 sub-groups, a smaller mixer 16-8-2 can exist without problems, the micro merely ignoring the lack of input sub-group channels until the mixer is expanded (if ever).

SOUNDTRACS
262A EASTERN PARKWAY
FARMINGDALE, NY 11735
(516) 249-3660

For additional information circle #74

MICMIX DC-2 REVERB DECAY CONTROL AND NOISE REDUCTION SYSTEM

The new MasterRoom DC-2 is said to allow the user to vary the decay time of virtually any reverberation system, while reducing noise generated by reverbs by up to 30 dB, and can be utilized on plates, live chambers, spring systems, or virtually any other type of reverb device.

The unit incorporates patent-pending downward expansion technology that is said to provide a smooth decay at any control setting. The DC-2 does not alter the tone quality of the reverb, as is the case with a damping system, and can shorten decay times by up to 75%.

The DC-2 is a two-channel unit that can be used with a stereo reverb, or two totally independent monaural systems. Front panel controls include a continuously variable decay time adjustment control, and an in/out switch for each channel. The reference level is internally adjustable from -10, 0, +4, to +8 dBv.

Suggested user price of the MasterRoom DC-2 is $695.00.

MICMIX AUDIO PRODUCTS, INC.
2995 LADYBIRD LANE
DALLAS, TX 75220
(214) 352-3811

For additional information circle #75

AUDIO KINETICS INTRODUCES Q.SOFT SOFTWARE OPTION 64

Designed for film or video post-production, the new Q.Soft Option 64 operating software may be installed in any Q.Lock 3.10 synchronizer to allow

For additional information circle #76

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 #77

New Products

users to customize operating procedures for specific applications.

Option 64 more than doubles the memory capacity of the Q.Lock, making possible the combination of both ADR (automatic dialog replacement) and SFX (sound effects assembly) programs. The operator may now select either of these special operating procedures, or a standard program directly from the Q.Lock controller.

With eight "Menu Pages," each allowing eight options, the user can further select or de-select up to 64 optional operating routines. For example, a programmed record loop may be executed in a rehearse mode with machine input switching, and then a single button depression will repeat with record entry and exit on an assigned machine.

AUDIO KINETICS
4721 LAUREL CANYON BLVD.
NORTH HOLLYWOOD, CA 91607
(213) 980-5717

For additional information circle #79

AURATONE INTRODUCES FIVE NEW MONITOR SPEAKERS

The new models include the T5 Ultra-Compact Two-Way, T6 Sub-Compact Two-Way, T66 Compact Two-Way, QC66 Quality Control Three-Way, and RC66 Road Cube Two-Way, all of which feature polypropylene low-frequency drivers, wide dispersion dome midranges, tweeters and super tweeters.

All the units incorporate six- or 10-element crossover networks with precision metalized film polyester capacitors, and air-core inductors mounted on specially designed fiberglass/resin PCBs. With the exception of the RC66 Road Cube Two-Way, all the new Auratone monitors are produced in mirror-image pairs for enhanced stereo imaging.

Enclosures are manufactured from low resonance Super-Acousticwood™, a high-density, wood-based product with acoustic properties that are said to be superior to the "particle board" used by nearly all other speaker manufacturers.

Suggested pro-net prices per pair: T5 \$250; T6 \$325; T66 \$595; QC66 \$695; and RC66 \$795.

AURATONE CORPORATION
P.O. BOX 698
CORONADO, CA 92118
(619) 297-2820

For additional information circle #80

MODEL 2711X DIGITAL EQUALIZER/RTA FROM NEPTUNE

The new 2711X combines the functions of the Model 2709B RTA with the 2711 27-band graphic equalizer, and adds a microprocessor to enable gain settings on each of 27 ISO-centered filtering bands to be stored and recalled in as many as 27 memory units. The storage of EQ settings means that many specific requirements for EQ can be recalled on demand, by the push of a button.

Additionally, the 2711X incorporates an RTA that can display an audio spectrum on the unit's 27-band LED matrix to locate room acoustic anomolies. The microprocessor also can perform other useful functions with the RTA, including triggering the RTA sampling from the audio source, so that the transient spectral qualities of the source can be frozen on the display. This feature allows more detailed EQ action where slight phase effects can degrade the sound.

Front-panel controls include a four-character alpha-numeric display that lets the user know at all times what function the 2711X is performing. Additional single LEDs give status information, such as which of the inputs is selected. Selection of functions is interactively programmed with only four switches, via the four-character display.

The 2711X can accept option modules for additional functions, or off-line connection to computer equipment, such as a printer for archiving data from the RTA section.

NEPTUNE ELECTRONICS
934 NE 25TH AVENUE
PORTLAND, OR 97232
(503) 232-4445

For additional information circle #81

YAMAHA INTRODUCES PRODUCER SERIES MULTITRACK CASSETTE

The M44 Mulitrack Cassette permits independent recording on four tracks with full overdubbing, in a compact package weighing just 12 pounds, and measuring 12 by 12 by four inches.

Independent record level controls are provided for each channel, and a choice of both Dolby B and C noise reduction is included. Zero stop, and zero start functions are featured, along with full logic transport controls.

When combined with the Yamaha MM30 Portable Mixer, the MT44 forms the heart of an easy-to-use multitrack recording and playback system; the companion RB30 system rack and patchbay, have been built specifically to accommodate the two components.

The MT44 and RB30 have suggested retail prices of $570 and $175, respectively.

YAMAHA COMBO PRODUCTS
P.O. BOX 6600
BUENA PARK, CA 90622
(714) 522-9134

For additional information circle #82

MULTI-CHANNEL D/A AND S-100 BOARDS FROM ICD

The D/A64-100 produces 64 analog outputs with 8-bit converter resolution, while the sister A/D 64-100 board performs analog-to-digital conversion with similar accuracy. Voltages may be generated or read over a 0-to-5 VDC range in 255 increments. The S-100 bus boards are port-selectable, so that multiple cards may be used to create large systems as controllers for various systems in the studio environment.

The CCA-100 Calendar/Clock/Alarm Board can be used to display hours/minutes/seconds, and day/month/year on a CRT; to time events in second increments; and to produce musical alarm tones over a four-octave range. Its companion CCT-100 Calendar/Clock/-Timer Board can control events with 0.01-second accuracy, keep track of computer time used, or calculate days elapsed between dates, all as hardware functions. Time/Date information may be sent to a printer or stored as data, with all functions under software control. Both cards have long term battery

backup, and utilize a minimal number of Z80/8080 ports for operation.

INDUSTRIAL COMPUTER DESIGNS
31121 VIA COLINAS #1005
WESTLAKE VILLAGE, CA 91362
(213) 889-3179

For additional information circle #83

For additional information circle #84

New Products

STAGE SYSTEM 200 FROM ELECTRO-VOICE

Housed in a black plastic cabinet with gray grille, the 36-pound, 1.8-cubic-foot Stage System 200 is said to be capable of producing sound pressure levels in excess of 120 dB at 1 meter on axis. Computer-optimization of coverage angle, woofer size, and crossover frequency results in a Controlled Directivity system that provides a well-defined 100-degree horizontal and vertical coverage zone of acoustic output in the critical frequency range of 500 Hz to 10 kHz.

The portable stage speaker system employs a high-output version of EV's Super-Dome tweeter, coupled to a high-frequency Direktor™ that is molded as an integral part of the cabinet. This coupling of a direct radiator to a directivity-controlling device duplicates the performance of an EV constant directivity horn with similar high efficiency, the company claims.

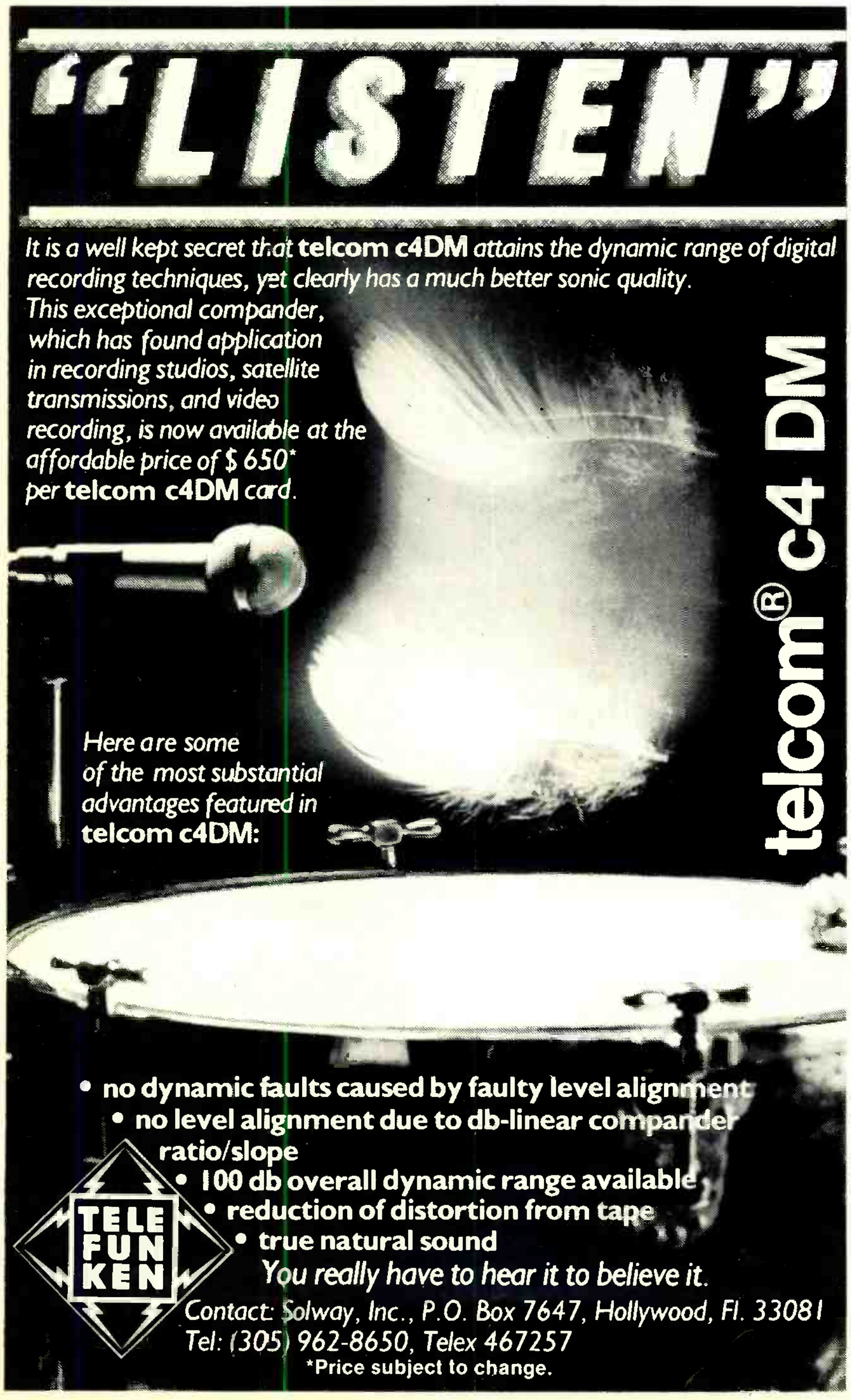
The low-frequency section boasts the recently developed EVM Pro-Line Model 12S, which can handle 300 watts continuous power (per EIA Standard RS-426). The unit's crossover network is a 12 dB per octave dual section type, with the crossover point at 2 kHz.

The Stage System 200 has a suggested list price of $589 ($491, pro net); the

optional S-200 active equalizer is available for $149 ($124, pro net).

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #86

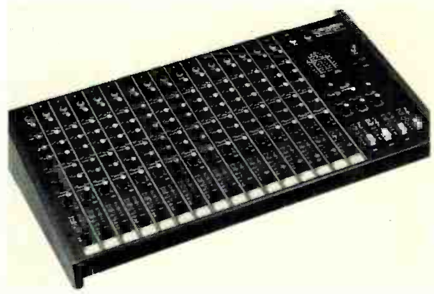


For additional information circle #85

ELECTRO-VOICE EXPANDS SERIES 52 MIXER LINE

Additions to the Series are the eight-channel EVT 5208, and the 16-channel EVT 5216, both of which employ individual plug-in printed circuit boards for each channel.

Each mixer input channel accepts a balanced low-impedance mike- or unbalanced high-Z line-level source. A channel effects insert is provided on each input, as well as on the two sub-group outputs. The three-band EQ section offers ±15 dB at 100 Hz, ±12 dB at 3 kHz, and ±15 dB at 10 kHz. Each chan-



nel also features an effects/reverb send, monitor send, pan control, peak LED, and channel fader.

The EVT 5208 and 5216 have pro net prices of $825 and $1,275, respectively.

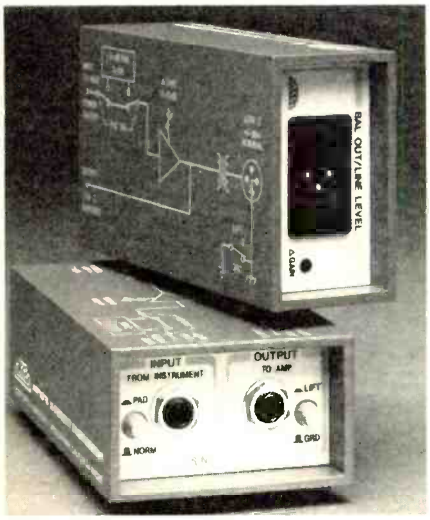
ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #87

DI-100 DIRECT BOX WITH ADJUSTABLE GAIN FROM AXE

The DI-100 utilizes a low noise BI-FET buffering amp at the instrument-input jack to eliminate the effects of loading, and provides a low-impedance signal to the instrument amplifier allowing for long cable runs. The XLR output is transformer balanced from the buffer pre-amp stage, and utilizes a special line-level output Jensen transformer.

Adjustable gain allows an engineer to optimize the instrument's level to the console for maximum signal-to-noise



ratio. Unlike other direct boxes on the market, which actually lower the instrument's output level at the XLR output, the DI-100 is said to send a low-impedance line-level (+4 dBm) signal, thus eliminating the necessity to use the console's mike pre-amp.

The unit can be powered either by its internal battery, or phantom power from a console. The unit is completely EMI/RFI shielded, and can send a +18 dBV signal to the console on 48V phantom power.

ARTISTS X-PONENT ENGINEERING
P.O BOX 2331
MENLO PARK, CA 94025
(415) 365-5243

For additional information circle #88

VALLEY PEOPLE 430 SERIES SIGNAL PROCESSORS

Three models of the new 430 Series are configured as successors to the 410/420 Dyna-Mite™ and Dyna-Mic multi-function signal processors.

The Model 430 consists of two channels of Dyna-Mite processing. Each channel is individually capable of performing limiting, expanding, noise-gating, keying, FM limiting, de-essing and voice-over ducking. Included in each channel is Valley People's Linear Integration Detector, which is said to yield flatter VU meter readings in the limiting mode than can be achieved by devices using Peak or RMS detection schemes. As a limiter, Threshold/Output Coupling is also offered to maintain a predetermined output level, regardless of the amount of limiting. An on-board "Anticipatory Release Computer" ensures short release times without excess pumping and modulation distortion. The two channels may be coupled for stereo operation.

The Model 431 is a combination of one Dyna-Mite and one Dyna-Mic channel, while the Model 432 comprises two channels of Dyna-Mic. The Dyna-Mic employs a modified Trans-Amp™ transformerless pre-amplifier, and each of the two independent sections will accept a variety of input levels. Either or both inputs may be passed through the 3-band EQ section, offering a quoted 14 dB cut/boost in each band.

VALLEY PEOPLE, INC.
P.O. BOX 40306
NASHVILLE, TN 37204
(615) 383-4737

For additional information circle #89

MC 734 CONDENSER MIKE FROM BEYER DYNAMIC

The new vocal condenser microphone is said to feature a flat frequency response from 20 Hz to 18 kHz, and is equipped with a special 3-step filter to compensate for the proximity effect in close-miking situations. A built-in stage resonance filter is said to suppress hissing and pop, rumble, and handling noise.

The MC 734 may withstand sound pressure levels as high as 138 dB at 1 kHz, and can be fed by any 48 volt phantom power source.

BEYER DYNAMIC, INC.
5-05 BURNS AVENUE
HICKSVILLE, NY 11801
(516) 935-8000

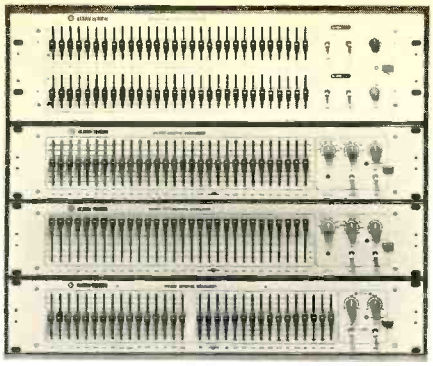
For additional information circle #90

KLARK-TEKNIK SERIES 300 GRAPHIC EQUALIZERS

The DN 300 third-octave, single-channel graphic equalizer has 30 control faders, each giving ±12 dB of boost and cut. It also incorporates adjustable high- and low-cut shelving filters, with selectable 6 or 12 dB per octave high cut slope. (Low cut is fixed at 12 dB/octave.) Electronic input balancing is standard, as are automatic failsafe bypass, and a manual equalizer override.

The DN 301 third- octave attenuating

equalizer — launched earlier in the year — has been specially designed for the sound contractor who needs to optimize overall sound levels in a particular installation. The 30 ISO center frequencies exactly match the display of the company's DN 60 spectrum analyzer. Features are similar to the DN 300 except for the fader control, which is 15 dB cut only at each of the ISO centers.



DN 332 is a dual-channel two-third-octave graphic equalizer with 32 control faders, each giving ±12 dB of boost and cut. A subsonic filter is incorporated in each channel, and combined with the equalizer bypass switch. Electronic input balancing and LED overload indication are standard.

DN 360 dual-channel third octave graphic equalizer is a development of the company's DN 30/30. The new version incorporates newly designed proprietory filter circuits, and also has improved front-panel graphics. The unit features 60 control faders, each giving ±6/12 dB boost or cut, and has a switchable subsonic filter in each channel.

KLARK-TEKNIK ELECTRONICS
262A EASTERN PARKWAY
FARMINGDALE, NY 11735
(516) 249-3660

For additional information circle #93

For additional information circle #92

DRUMULATOR DRUM MACHINE FROM E-MU SYSTEMS

The Drumulator's internal, digitally recorded sounds include bass drum, snare, rim, three toms, clave', hand claps, cowbell, open and closed hi-hat and ride cymbal (which can be replaced with an optional crash cymbal).

Rhythm patterns can be programmed in any time signature, and then combined into complete songs with programmable control of levels, repeats, and tempo (including programmed tempo changes within a song).

The unit contains a programmable mixer that allows a different mix to be stored and instantly recalled for each song. These mixes can even be programmed to change in the middle of a song. A programmable accent facility provides access to a regular and accented version of every sound, with the accent level independently user selectable for each drum, cymbal and perscussion instrument.

Other features include the ability to sync to tape or other sequencers, programmable trigger output, and a built-in computer interface.

The Drumulator's suggested list price is $995.

E-MU SYSTEMS, INC.
2815 CHANTICLEER
SANTA CRUZ, CA 95062
(408) 476-4424

For additional information circle #94

MODEL 234 SYNCASET FOUR-TRACK CASSETTE DECK FROM TASCAM

The Model 234 Syncaset™ is claimed to be the world's first professional 4-track cassette recorder/reproducer. The Model 234 records at 3¾ IPS, is equipped with dbx noise reduction, and utilizes a 3-motor transport with servo-controlled capstan that is said to keep wow and flutter to 0.04%. Pitch control of ±12% in record and play enables varying the speed for special effects and tuning.

Each track can be separately placed in record ready or record mode. An Input/Sync switch allows monitoring of either the input for a track or the prerecorded track up to the drop-in point. The Line and Mic/Instrument inputs of each channel are internally mixed, allowing recording of up to eight sources on four tracks in a single take.

The rack-mountable Model 234 was designed to match with a mixer to add EQ, cue and monitor mixes, solo, effects sends/returns so the two become a full-scale portable 4-track studio.

TASCAM/TEAC
7733 TELEGRAPH ROAD
MONTEBELLO, CA 90640
(213) 726-0303

For additional information circle #95

LEXICON MODEL 200 DIGITAL REVERB

The new Model 200 Digital Reverb and Room Simulator was designed to provide cost-effective processing in small studio, live performance, and broadcast applications.

Features include continuously variable predelay, reverb time from 0.2 to 70 seconds, and acoustic ambience controls intended to simulate the spatial characteristics of acoustic environments from a phone booth to the Grand Canyon.

Preset effects include several plate reverbs, chamber simulations, and concert hall simulations. Non-volatile memory allows any setup to be stored and recalled on demand even if the unit has been turned off in the interim or power interrupted. Controls adjust reverb frequency contour, high-frequency rolloff, echo density, and early reflections, creating a wide variety of sonic characteristics.

Frequency response is quoted as being within ±0.5 dB from 20 Hz to 10 kHz; 84 dB is typical dynamic range, with 81 dB minimum, 20 Hz to 20 kHz noise bandwidth. Noise and distortion are said to be 0.04% typical, 0.07% maximum, at 1 kHz reference level. Input level range is -12 to +24 dBm, and outputs are capable of driving all standard lines and downstream devices.

The Lexicon Model 200 occupies 5¼ inches of rack space and weighs 18

New Products

pounds. A remote-control jack permits selection of preset programs by footswitch or external logic signal control.

LEXICON, INC.
60 TURNER STREET
WALTHAM, MA 02154
(617) 891-6790

For additional information circle #98

OBERHEIM UNVEILS MODEL 700 STEREO AMP

The new Model 700, while rated at 200 watts RMS per channel into 8 ohms, and 350 watts RMS per channel into 4 ohms, can provide musical peaks of 250 and 440 watts respectively. THD is quoted at 0.1% from 20 Hz to 20 kHz at full rated output; power bandwidth extends from 10 Hz to 40 kHz.

The unit is said to have been extensively tested for reliability under exceedingly adverse conditions. The full output of one channel (150 Vpp) was fed directly into the other channel's input, proving that low-level input circuits could not be harmed by high-voltage sources. In other tests, with both channels driven to clipping, all types of loads were applied, ranging from open and short circuits to highly reactive loads. Since the amplifier uses no VI limiter, no distortion resulted when driving highly reactive loads.

Hum and noise are quoted at 104 dB below rated output, and the unit operates at rated loads without the need for noise-producing fans.

OBERHEIM ELECTRONICS, INC.
2250 S. BARRINGTON
LOS ANGELES, CA 90064
(213) 473-6574

For additional information circle #99

MDM-TA2 TIME-ALIGN NEARFIELD MONITOR FROM CALIBRATION STANDARD INSTRUMENTS

The new MDM-TA2 Nearfield™ Monitor features a "Position/Program" switch on the front panel that adjusts the response for listening position (NFM or Distant), and program material (Original or Final), so that proper equalization may be applied to the original recording to overcome upper-range losses in the signal chain.

The monitor is said to be the first unit to provide a Polarity switch on the front panel, to allow the absolute polarity of program material to be checked easily.

Response is a quoted ±3 dB, 60 Hz to 20 kHz, and the monitor can produce 108 dB SPL at 1 meter. The MDM-TA2 is sold in pairs matched to ±0.5 dB.

CALIBRATION STANDARD INSTRUMENTS
P.O. BOX 2727
OAKLAND, CA 94602
(415) 531-8725

For additional information circle #100

ELECTRO-VOICE ADDS THIRD SENTRY TO MONITOR LINE

The new Sentry 505 studio monitor features an angled enclosure for ceiling/wall mounting locations, and is an acoustic match for the larger Sentry 500 monitor. The smaller cabinet volume of the 505 has been carefully calculated to roll off the system's low frequency at a rate that compensates for the bass boost which occurs when a speaker system is mounted in a "quarter space" environment (where the speaker is mounted at the intersection of two large surfaces

For additional information circle #101

such as a ceiling and wall).

The 505 is capable of producing 96 dB SPL (1 watt, 1 meter, anechoic), and features frequency response that is essentially flat from 40 Hz to 18 kHz. While it can be powered by modestly-sized amplifiers, the unit will handle 100 watts average long term, and short term peak loads of 400 watts.

A two-way speaker system with a 12-inch, high excursion woofer and a tweeter capable of handling a full 25 watts, the 505 has a crossover frequency of 1.5 kHz. A specially designed "director" for the tweeter matches the dispersion angles of both transducers at the crossover point. The result is well controlled vertical and horizontal dispersion of sound in the critical 250 Hz to 10 kHz range, Electro-Voice claims.

ELECTRO-VOICE, INC.
600 CECIL STREET
BUCHANAN, MI 49107
(616) 695-6831

For additional information circle #102

"GREAT BRITISH SPRING" STEREO REVERB UNIT FROM CONNECTRONICS

In effort to achieve a natural sound, fancy electronics have been avoided in the Great British Spring stereo spring

reverberation unit. Twelve springs, closely matched to complement each other, are employed in the design, and are said to produce a full, smooth effect that is virtually flutter-free. Varying lengths, mass, and coil diameters ensure wide diffusion of the reverberant energy.

The unit is available in three configurations: unbalanced with ¼-inch stereo jack inputs and outputs; unbalanced but with XLR connectors; and balanced with XLR connectors.

Suggested professional user price is $495.00 for the unbalanced jack input/output version and $687.50 for the balanced XLR version.

CONNECTRONICS CORP.
652 GLENBROOK ROAD
STAMFORD, CT 06906
(203) 324-2889

For additional information circle #103

MODEL SH-1 STEREO POWER AMPLIFIER FROM SESCOM

The Model SLH-1 was designed to provide an economical means of powering small control room monitors or, used in conjunction with Sescom's SHB-1 Headphone Junction Box, for powering multiple headphones.

The unit features a power output capability at 8 ohms of 20 watts per channel, both channels driven, with less than 0.1% at 1 kHz for maximum rated

output.

Manufacturer's suggested retail price is $160.

SESCOM, INC.
1111 LAS VEGAS BLVD. NORTH
LAS VEGAS, NV 89101
(702) 384-0993

For additional information circle #104

New Products

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The new condenser omicrophone is equipped with a rectangular dual membrane capsule, and is said to have low inherent noise with superb front to back ratio.

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CAMERA MART INC.
456 W 55TH STREET
NEW YORK, NY 10019
(212) 757-6977

For additional information circle #107

B&W DM110/220 MONITORS AVAILABLE FROM ANGLO AMERICAN AUDIO

Essentially, the new DM110 and DM220 Digital Monitors are similar in concept; the DM110 is a two-way vented design employing two drive units; while the MD220 is a sealed enclosure, employing three drivers — a pair of 8-inch drivers for bass and midrange, and a one-inch dome tweeter — with greater power handling capability, and an extension in bass response.

Design brief for both models was high sensitivity of not less than 90 dB, 1 watt at 1 meter; a broad, extended, and linear frequency response; and a dramatic

reduction in production cost, with totally re-designed drive units. In addition, due to good polar distribution and extremely close tolerances between pairs, stereo imagery and depth information from digital material are said to be accurately interpreted.

Along with their increased sensitivity, both the DM110 and DM220 are capable of high acoustical outputs. Typically, a pair of DM220s can produce peak listneing levels higher than 115 dB SPL in a medium-size listening environment.

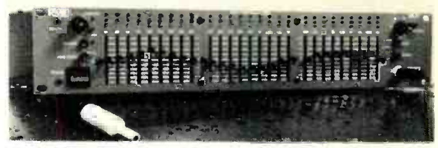
Suggested list price for the DM110 is $149, and the DM220 $249.

ANGLO AMERICAN AUDIO CO.
1200 MARKHAM ROAD
SCARBOROUGH, ONTARIO
CANADA M1H 3C3
(416) 438-1012

For additional information circle #108

RANE RE27 REALTIME EQUALIZER

The RE27 combines a precision third-octave equalizer with a third-octave realtime analyzer, both housed in a single rack-mount chassis. The analyzer display consists of three LEDs above each equalizer slider: red, green and yellow. After activating the built-in pink noise generator, one simply moves each slider up or down until the green LED above it is lit; when all LEDs are green, the system is normalized to within ±1 or ±3 dB accuracy, switchable.



Additional features include calibrated flat-response condenser microphone with cable; switchable curve select for "Flat" or "House Curve" (for smaller clubs and lounges); ±1 dB or ±3 dB select for display window; 0 dB adjust with calibrated SPL readings; LED-indicated hard-wire bypass for the equalizer section, and transformerless auto unbalanced/balanced/floating input and outputs.

suggested list price for the complete system is $799.

RANE CORPORATION
6510 216TH SW
MOUNTLAKE TERRACE
WA 98043
(206) 774-7309

For additional information circle #109

NEPTUNE ADDS TO LINE OF POWERED MIXERS

The new 411P, 611P, and 811P powered mixers, equipped 4, 6 or 8 input channels, offer many of the features found on NEI's XM and '22 series consoles, including electronically balanced mike inputs; monitor and reverb/effects sends; and high and low input channel equalizer.

The 411P and 611P feature a built-in amplifier, FTC rated at 80 watts RMS at 8 ohms, and 125 watts RMS at 4 ohms. The 811P contains two of these amps, which also can be mono-bridged to produce over 250 watts RMS into 4 ohms.

Both the 611P and 811P include a full octave, 10-band graphic equalizer on standard ISO centers, which is connected to the main output but can be externally patched to the monitor send, or any other external device.

NEPTUNE ELECTRONICS
934 N.E. 25TH AVENUE
PORTLAND, OR 97232
(503) 232-4445

For additional information circle #110

NEW REFERENCE MONITOR FROM KEF

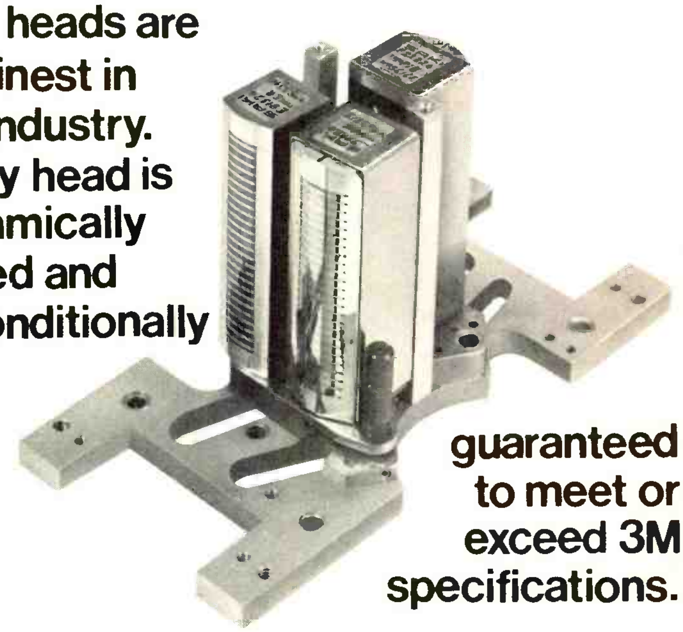
The new KEF KM-1 monitor loudspeaker features an integral power amplifier with a maximum output exceeding 1,200 watts, and which is comprised of two power supplies and eight separate output sections to feed the unit's seven drive units. An active three-way dividing/equalizing network is included, fed from a separate power supply. The new loudspeaker is capable of delivering 120 dB SPL on program peaks under typical listing conditions; with frequency response of 30 Hz to 20 kHz ±2 dB; and a signal-to-noise ratio of over 100 dB.

In keeping with KEF's design philosophy, the new KM-1 features an S-type soft clipping limiter, which is automatically activated under conditions of near continuous peak overloading. (This is said to permit up to a 6 dB increase in loudness without significant audible distortion.) Additionally, full electronic overload protection safeguards all the drive units and integral electronic circuits against accidental overloads.

Prototypes of the KEF KM-1 have been in use at the British Broadcasting Corporation's Maida Vale music studios, London, for more than a year, and now is in general production.

KEF
425 SHERMAN AVENUE
PALO ALTO, CA 94306
(415) 321-2035

For additional information circle #111



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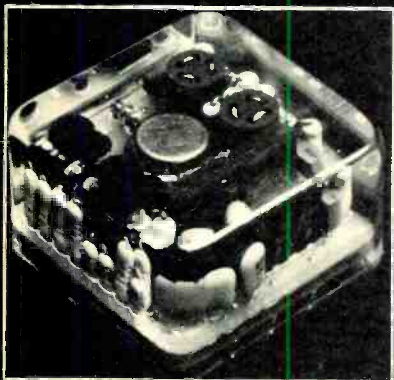

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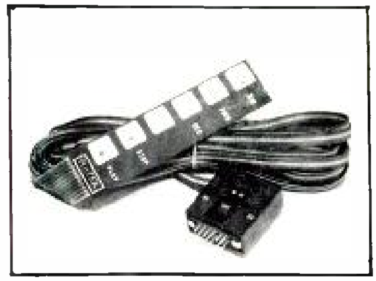

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For additional information circle #121

INDUSTRY INVENTIVENESS

. . . continued from page 24 —

drama (a group therapy method where members act out traumatic moments in their past). A few assistant engineering jobs later Gazecki wound up as first engineer at Producer's Workshop. Shortly after this he was called on to help design and install a studio at Elektra-Asylum, and this led to his working with Paul Rothchild. Gazecki and Rothchild hit it off so well that Bill was asked to work on several other projects, including some film work at 20th Century Fox. Becoming associate producer with Rothchild on *The Rose* soundtrack, Gazecki was exposed to the world of film sound production.

"Film sound deals with effects, dialog, and music, and pretty much requires *team* mixing. When you go into film or TV after having been in records, you have to learn to relate to a project as more of a collective effort. Mixing for film is a very different kind of mixing. In music, there's a subconscious tuning-in that allows an engineer who has rehearsed a song extensively to sense the rhythms, but there's nothing like that in film or television. You have to start relating to timecode, or footage and frames, instead of rhythms or bars of music."

Once he got a taste of film and television sessions, Gazecki's interest grew. He took a position with Ed Lever's Canyon Recorders in West Los Angeles, ending up being put charge of audio sweetening for the *You Asked For It* television program. "My job was to put together the music, the dialog, and the sound effects, so that it all sounds congruous with the picture," he recalls. "Most of the sound for this type of show is done in post-production, so that means a *lot* of dubbing."

Gazecki says that many engineers from the music business are naturally looked upon as outsiders when they try to break into other kinds of recording work. Often that can mean a demotion in status, he says.

"Here I was working on *The Rose* with a pretty good name, but when I wanted to get into television I took a job as an assistant mixer. I was a mixer, but was working under another mixer and had *no* authority whatsoever. I had to mix according to their parameters. Just because I had mixed music did not mean to my employers that I could mix television. I accepted that because I wanted to learn."

Once a former music mixer plugs into the mechanics of television or film production, he can then draw upon more of his former experience, and advance relatively quickly. "Because records are a sound-only medium that involves the storage of music, and because the competition in so intense in the record business, you have to become quite highly trained to succeed in it. You have to really know sound and the technology involved; you have to know your multitrack systems, console automation, and all the characteristics of recording and acoustics.

"In audio-visual mediums, like film and television, sound isn't the primary factor involved, and therefore the training for sound people in those fields isn't as thorough as it is in the record business. If you can get over the different attitudes and environments between the two mediums, you can progress more quickly because you have the training to understand more of the process."

There are several differences in attitudes and environments between the "romance" of the music world, and the work-a-day attitude of television and film sound. Gazecki elaborates: "In music you relate much more personally to the artist and the final outcome of the project. In television and film, the sound guys are one of 20 departments and, with the exception of the production mixer who actually mikes the talent, you have almost no contact with the principal talents of the film.

"Consequently, the attitudes are different. There is a lot less glamour, and it's a lot less personal. There will be people that must approve your work, and who you'll never even see. They are somewhere down the line, but if they don't like your work you'll have to remix it, even if it has been six months. There's no room for argument or your opinion, which you can express in the music recording situation.

"This is offset, however, by the joy you experience in adding to a larger project. I get more personal satisfaction by being a contributor to a greater work, which can be much more emotionally effective."

One of the most interesting projects Gazecki has been involved in occurred while he was working at Canyon on the facility's digital sound effects library.

"Existing film sound effects libraries are over 10 years old," he says, "and there have been tremendous breakthroughs in technology since they were recorded. Digital recording is, of course, an incredible improvement in the quality, but there have also been breakthroughs in microphone technology, recording tape, miniaturization, and other areas. When the digital sound effects libraries are completed there will be a huge jump in quality."

Gazecki occasionally still works in the music recording medium, and recently teamed with Paul Rothchild again to work on a Doors live album assembled from tapes initially recorded at concerts in 1969 and 1970.

"The hardest thing was locating the tapes," he recalls. "All that existed were quarter-inch rough mixes — 70 rolls of eight-track master tapes had been lost in a 10-story warehouse. It literally came down to finding a tiny slip of paper in the bottom drawer of an abandoned desk that once belonged to a former supervisor of the warehouse. It took six months of lawyers and everyone saying someone else had the tapes, before this slip of paper was found. It's turning out to be a great record."

Another area of multimedia sound that is proving exciting to Gazecki is that the structures always change somewhat, because the nature of the project can be so different. "The requirements for *Star Wars* are much different than the requirements for *That Championship Season*, for example. There's different kinds of locations, different lighting, different feels, and each new project has an excitement of its own."

There are a large number of careers open to music engineers in the film and television mediums. Gazecki has some interesting things to say about the working structure of these industries.

"Sound for television doesn't go through nearly as many steps as the sound for film," he offers. "In television, basically you have three steps: initial recording; on-line editing, which is your master picture assembly; and video sweetening, which is where all the sound is added and conformed to the picture at one time. In film you have departments for music, effects, dialog dubbing, and editing.

"There may be less opportunities in television than film, because one person handles more functions. But in music it is even more limited, because one producer and engineer handle the entire project. Television is great training for film, because you learn about sound effects, music and dialog in relation to picture. "In both film and television sound you will probably start out as a mixer — if you have that experience in the music business — but the only way to really learn how to work in film and television is to jump in.

"Once you're working in the medium you realize that there's a world of careers open to you. You start realizing that you're in the 'entertainment business,' which has a much larger scope than just music and records. When you're an engineer working in the recording studio, you learn what the producer does and if you could do it. In film it's the same process, and yu can even go beyond all the various jobs there are in the sound field. You could be a lighting director; a post-production supervisor; a production sound mixer, where live recording is your speciality; a sound-effects editor; an optical transfer man; or a variety of other fields.

"The point is that once you work on films for a period of time, you start to learn how to do other things as well; you develop interests and learn other skills.

For example, you see what the lighting director does to give the director a certain kind of shot. You realize that you are in a business with a lot more career opportunities than you ever realized in the music business."

Gazecki has combined all his various skills into his current role as an independant sound consultant. "Because of the experience I've had, I'm now called into solve complex problems in film and television that aren't usually handled by just one person," he confides. "The reason these problems usually develop is because there are 'too many cooks spoiling the broth,' and communication gets confused. They need someone to see through the problems of each department."

Currently Gazecki is involved in *Bravisimo*, a Latin-produced prime-time television series, for which he recorded a band called Tierra for the show, and an album to be released later. Gazecki says that the show's purpose is to showcase major Latin musical talents, including Amanda, Miguel, and Yomo Toro, to a wider, networked television audience. His is a classic example of the kind of positive career change one can make if they have learned their craft well, and are unafraid to try and apply them in a new environment.

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news

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AUDIOTECHNIQUES TO SPONSOR DIGITAL SEMINAR

A seminar workshop on use of the Sony PCM F-1 digital audio processor, sponsored by Audiotechniques, Inc., will be held on Wednesday, September 28, at RCA's Studio A, 1133 Avenue of the Americas, New York City.

Goal of the seminar is to examine the various aspects of the PCM F-1 as a professional recording tool, and to provide an understanding of the advantages and limitations of this widely discussed recording medium. A number of digital recording specialists with experience in use of the F-1 will present a practical, "How-To-Do-It" course that will cover studio and remote recording, assembly and editing, plus mastering for analog and Compact Disc.

Seminar leaders will include Tom Steele of Frankford Wayne Mastering, Gus Skinas of Sony Digital, David Smith of Editel, and producer-engineer Tom Jung.

Fee for the seminar, which runs from 9:30 a.m. until 4:30 p.m., is $60.00; additional information from Ham Brosious, on (203) 359-2312.

INDUSTRIAL COMPUTER DESIGNS DEVELOPING COMPUTER-BASED SYSTEMS FOR PRO-AUDIO INDUSTRY

Headed up by Michael V. Ragsdale, founder and former president of Uni-Sync, Inc., the new company has created a line of over 50 computerized devices surrounding the concept of, according to Ragsdale, "physical computerization," or the use of computing hardware and software to monitor and control devices and tasks in "real-world" applications.

Several of the new ICD products have been created specifically for use as "building blocks" by the pro-audio industry, including a digitally controlled amplifier/attenuator, a control-signal multiplexer, and a computer capable of digitizing a large number of control voltages at high speed, while simultaneously performing machine synchronization, level set, signal-path routing, video control, time and track assignment logging, and a slew of other audio- and control-signal functions.

Ragsdale has a long history of involvement with the professional audio industry. Having set up, in 1968, his first company, Associated Sound, he then formed Uni-Sync, Inc., four years later to build mixing consoles, power amplifiers, electronic crossovers, equalizers, and other related products. Following the sale of the company to BSR, Inc., in 1976, Ragsdale went on to form Audio Logic, which produced the first multitchannel peak level monitor with auto-scaling for professional power amplifier applications. Audio Logic was sold in 1980 to Quantum Audio Labs, Inc. and, following design consulting work for UREI, Ragsdale formed ICD, which currently specializes in the design of microprocessor-based products, and performs design consulting for various hardware manufacturers in the computer, audio, and video fields.

PARDON OUR SLIP!

It would appear that the gremlins struck during preparation of the December 1982 *R-e/p*. On page 83 of that issue, mid-way through an eloquent treatise by Bob Hodas on the acoustic design of Tres Virgos studio, we inadvertently referred to the facility using carpeting on one secion of the studio floor, and "parkay" on another.

Now, while there may be some virtue in investigating the acoustic properties of a semi-solid margarine floor, we cannot help but wonder at the way absorption and reflectivity coefficients might change with temperature, altering the floor's normally firm surface to one that would be decidedly liquid.

And what of the musicians? Would they have trouble keeping their balance on such a slippery floor? And imagine having to explain to your nearest and dearest why your ankles are covered in crusted grease after a long session.

Of course, we should have referred to the floor being covered in *parquet*. Our thanks to Sam Borgerson of Studer-Revox for his astute observations, and apologies to Kraft for the oversight.
—*Editor.*

THIS ISSUE OF *R-E/P* IS SPONSORED BY THE FOLLOWING LIST OF ADVERTISERS

news

VALLEY PEOPLE SETTING UP SPECIALIST PRO-AUDIO SALES DEPARTMENT

The new sales department, which has been established to deal with the needs of professionals involved in industrial audio installations and sound contracting, will be headed by Ray Updike. "Many companies do not fully understand the individual needs of these specialized markets," Updike says, "and therefore are not set up to provide information and/or products applicable to them. Since Valley People is in the business of selling to product users, as well as product manufacturers, we are equipped to provide engineering services, from assistance in applications to complete systems design and fabrication."

— News Notes —

Studer-Revox America has opened a new Northern California office, to be headed by **Fred Layn**, formerly of the New York office, to represent the company in Oregon, Washington, Idaho, Montana, and Wyoming. The new address is: 954 Hawthorne Drive, Walnut Creek, CA 94596. (415) 930-9866. Meanwhile, the company's Southern California office, located in Van Nuys, has added **Vencil Wells**, formerly general manager of Kendun/Artisan/Sierra Audio, as sales engineer . . . **Soundcraft Electronics, Ltd.** has formed a Canadian subsidiary — Soundcraft Electronics Canada, Inc. — to be headed up by **Richard Lasnier**, with **Jean Daoust** as general manager. The company address is: 1444 Hymus Boulevard, Dorval, Quebec H9P 1J6. (514) 685-1610 . . . **Restoration** has relocated to larger premises, thereby tripling its operational size, by occupying Units 11, 12, and 12A within its existing building at 15904 Strathern Street, Van Nuys, CA 91406. The phone number remains the same: (213) 994-6602 . . . The following company has been appointed U.S. distributor for **MILAB** microphones: **Camera Mart, Inc.,** 245 West 54th Street, New York, NY 10019. Phone: (212) 757-6977 . . . **AMEK Systems and Controls** has moved its expanding operation to larger premises, and now has adequate space for demonstration and sevice, due to accelerated demand for the company's line of concert-sound and recording consoles. **Tim Mungovan** and **Artie Toshner** continue to head up the service and sales team at the new address: 11540 Ventura Boulevard, Studio City, CA 91640. (213) 508-9788.

SCHUBERT SYSTEMS GROUP EXPANDING CONCERT-SOUND SERVICES; ADDS DAVID MORGAN AS PARTNER

As of July 1, SSG took on Morgan as a partner, and expanded to a full-service sound reinforcement company by acquiring the assets of Innovative Audio, Inc. In the past, SSG electronics, monitors, and PA systems have been used by Toto, Willie Nelson, and the Doobie Brothers. This season the new company already is on the road with Willie Nelson, Paul Anka, Robert Palmer, and Christopher Cross.

The company's current complement of equipment includes eight Gamble consoles with proprietary SSG modifications, all-JBL biamped monitor systems, and full flying four-way JBL phase-array PA systems, all driven by SSG transformerless minimum-delay crossovers, and Cerwin-Vega Metron amplifiers. In addition, a full range of analog and digital effects is available, including two Lexicon 224X digital reverb units, with LARC remote control.

Managed by David Morgan, with Dirk Schubert as chief engineer, SSG is located at 157 E. 163rd Street, Gardena, CA 90248. (213) 532-4142.

PEN PAL

河 南 人 民 广 播 电 台

Recording Engineer Producer

先生们：

高兴地看到贵刊 对贵刊所登广告厂商产品很感兴趣 希望得到贵刊所登广告各厂家的产品详细资料及报价. 请费心予以转告他们. 尽快寄来. 以便我们进口设备时研究参考. 还有什么请一并予以推荐. 谢谢

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中华人民共和国 河南人民广播电台

宋井书 83.1.25

From the Editor:

Fortuitously, a few days after we had received the letter reproduced above, Mr. Jack Chen of Top Recording Studio in Taipei stopped into our offices, to renew their subscription, and order several of the books advertised in *R-e/p*. Notwithstanding differences in dialect (and no mention of differences in political attitude), Jack was able to translate the letter from an engineer at a radio station located in Hunan Province, of The People's Republic of China, which asks for additional information on the products advertised in *R-e/p*.

Should any manufacturer wish to respond, *R-e/p* will be delighted to send an accurate reproduction of the address in the original Chinese. This will be suitable for photocopying, and affixing to an envelope.

For the record, more than a dozen subscriptions have been purchased for delivery inside The People's Republic. Additionally, we are informed, the majority of a bulk shipment of *R-e/p* to a prominent Hong Kong audio/electronic distributor is destined for delivery to The PRC. ■■■

Studer's Secret of Success

In years past, the Studer A80VU has earned widespread acceptance by the world's premier recording studios. And this success story is far from over; top studios continue to choose the A80VU MKIII over other "all new" machines. The secret of this success lies in three basic rules:

1. If it can't be made better, don't change it.
2. If improvements can be made, make them – even if they don't show on the outside.
3. Use longer production runs to hold down final cost.

Now in its third generation, the Studer A80VU series once again offers state-of-the-art performance at a surprisingly modest price. The new A80VU MKIII system incorporates several significant improvements, including:

- **Transformerless Line Amps:** Low output impedance assures optimum performance even with long cable runs.
- **New Headblock:** Tight spacing of erase and record heads permits exceptionally accurate punch-in and punch-out.
- **Remote Unit:** Full channel remote with 20-address memory autolocator.

Instead of repackaging these changes in an "all new" machine, Studer kept the basic transport – a design with an unprecedented reputation for reliability. Also, because basic tooling costs have long since been amortized, the A80VU MKIII's price has been held down, thus offering a better price/performance ratio.

How much better? Call your nearest Studer representative for details. He'll help make our secret the key to your success.

STUDER REVOX

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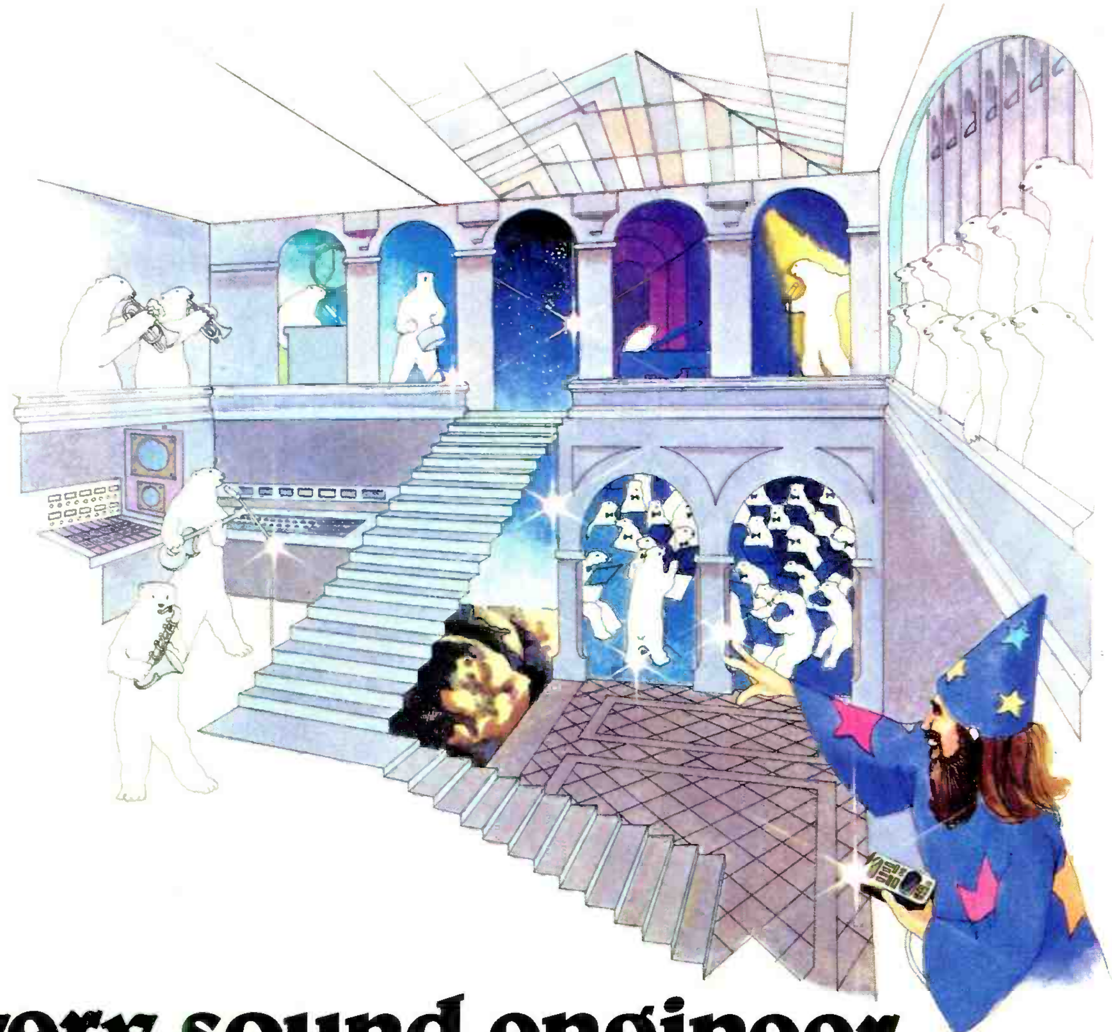

Every sound engineer needs a little magic

The Ursa Major 8X32 digital reverberator puts pure magic at your fingertips. Touch a button and you're transported to a concert hall stage. Touch another and hear sound roll through vast, empty canyons. And another, to conjure up a bright, tight plate that adds body to brass and drums. And then another: a larger plate, warm and beautifully balanced for voices.

In fact, the 8X32 lets you create and explore an almost infinite universe of acoustic environments. Four pre-set programs establish basic spatial qualities; then, the microprocessor-based controls allow you to separately fine tune all seven key reverberation parameters. For more control, there's a full remote console, LEDs that constantly display all the panel settings, and 64 registers of non-volatile memory to preserve and recall useful set-ups.

And the 8X32's acoustic spaces *sound real.* Rich, uncolored and clean—even with difficult material and decay times as long as 20 seconds.

If you'd like to add a little magic to your sound, spend some time with an 8X32. We invite you to write us for detailed system specifications, prices (surprisingly low), and the name of a local Ursa Major dealer. See for yourself what it's like to acquire powers you once thought existed only in your imagination.

The 8X32 DIGITAL REVERBERATOR

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