RECORDING ENGINEER/PRODUCER

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

$3.00
June 1984
Volume 15 — Number 3

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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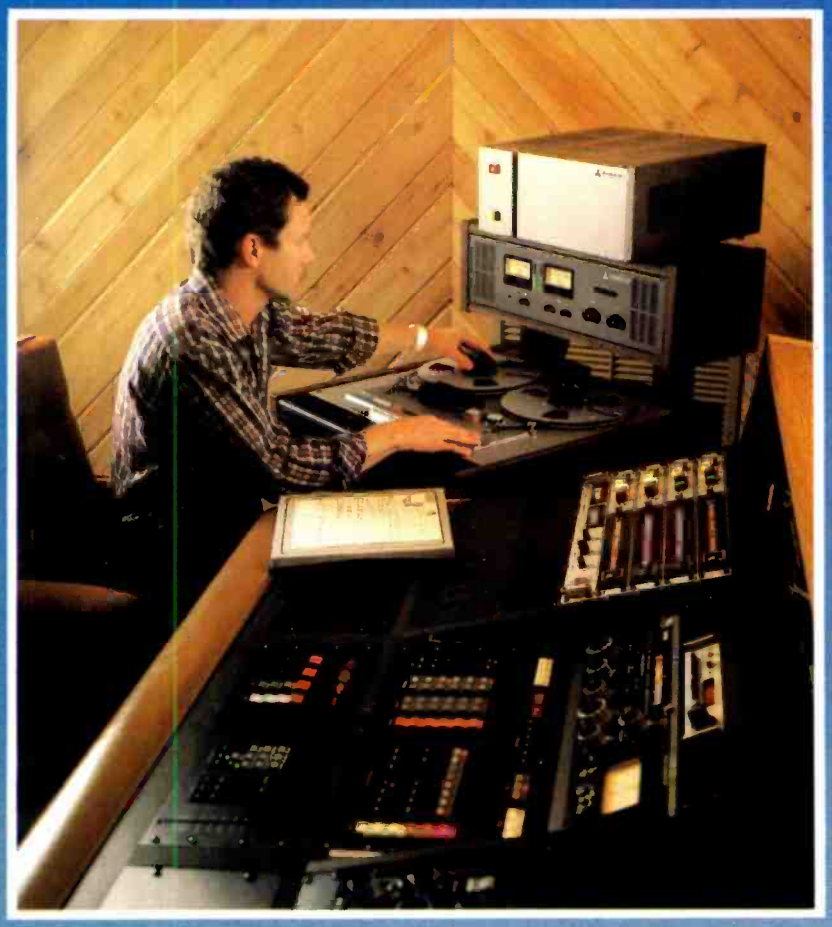
DIGITAL REALITY IN DISC MASTERING:



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FUTURE DISC SYSTEMS IN HOLLYWOOD.

Future Disc Cuts The Hits

Future Disc Systems was one of the first disc mastering facilities to purchase the Mitsubishi X-80 Digital Mastering System. When owners Steve Hall and Gary Rice first opened their doors, top artists began requesting that their lacquers and Compact Disc masters be produced from Mitsubishi X-80 masters. Their investment has paid off handsomely both in the quantity and the quality of their disc mastering business. A simple combination: World Class artists mixing to the Mitsubishi X-80 Digital Master Recorder and Future Disc cuts the hits.

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The Mitsubishi X-80 Format is fast becoming the World Class standard for disc mastering. And for good reasons. In New York, Los Angeles, Nashville, San Fransisco, and Miami, more X-80 masters are being produced than perhaps on any other digital system.

X-80 is Affordable

While the other professional digital audio 2-track recording systems require video tape recorders and complicated outboard systems costing as much as $100,000 (for a practical system), the reel-to-reel X-80 is available for about $25,000. And it is simple to operate as a regular tape recorder. And more reliable.

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The X-80 is regarded as the best sounding digital audio recorder in the business. One reason is the wider frequency band available compared to the video cassette based systems, yielding a natural and more desirable sound.

Easy Editing

The powerful error correcting system provides for click-free performance even after a conventional cut and splice operation! Yes, the X-80 is the only digital audio studio master machine on the market that allows you to do razor blade editing. No need for an expensive electronic editing system to perform simple edits. Cut, splice and listen. No need to play through an entire tape reel to perfect the edit point as you need to do with the other expensive video cassette based system. No special training is needed to operate the X-80. You use it just like an analog machine, but it's digital

Outstanding Performance

With a dynamic range in excess of 90 dB, the X-80 beats the ½" analog master recorders by more than 20 dB. That's a reduction in background noise level by a factor of 100. And you can make multiple generation copies where the last copy is as good as the first. Consider tape costs, both in the studio and in the propagation of master copies. The X-80 uses ¼" tape at 15 IPS (one hour record time on 10.5" reel) while the analog master recorders use ½" tape at 30 IPS. That's four times the tape consumption — with 20 dB less dynamic range (or worse). Believe it. Studio digital audio makes a lot of sense, both technically and financially.

Compact Disc from X-80

The X-80 is now the most convenient format for Compact Disc studio masters. Easy and inexpensive to edit, sequence, and make digital copies from, the X-80 masters are now being processed by PolyGram in Germany and Denon in Japan for Compact Disc mastering. We believe that you'll hear the difference the Mitsubishi X-80 Digital Mastering System makes. We invite you to call or write for complete details.

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MITSUBISHI DIGITAL AUDIO SYSTEMS

For additional information circle #3

RECORDING ENGINEER/PRODUCER

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

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

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

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June 1984
Volume 15 — Number 3

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Studio A at **Star Track,** a three-room facility based in Tulsa, and owned by Rod and Sally Slane. The studio specializes in custom jingle recording, scoring for industrial films and audio/video presentations, and demo sessions for regional and local artists. Control room equipment centers around an AMEK/TAC 16/8/2 console, augmented by a pair of 8MX monitor sections, and an 8CX add-on input section, providing a total of 24 input channels routing to 24 groups via eight subgroups. The board has also been modified to provide full 24-track monitoring, and in mixdown can accommodate 48 tape returns. Tape machines include an Otari MTR-90 multitrack, MTR-10 two-track, and an MX-5050 MkIII four-track/half-inch transport. A Roland Compu-Edit automation system provides VCA level control for up to 15 input channels, and is used in conjunction with an Adams-Smith Model 2600 SMPTE synchronizer for audio sweetening sessions.
Photography by *Gary Gibson.*

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Made in U.S.A.

The challenge to console manufacturers of the eighties is to design new mixing systems that match the dynamic range, distortion specs, and frequency response now possible on digital magnetic tape. AUDIOARTS/WHEATSTONE has taken this challenge and designed the 8X Recording and Production Console. Today, through careful engineering, the technical performance of the 8X is approaching all possible theoretical limits, resulting in the smoothest, most transparent console we have ever built.

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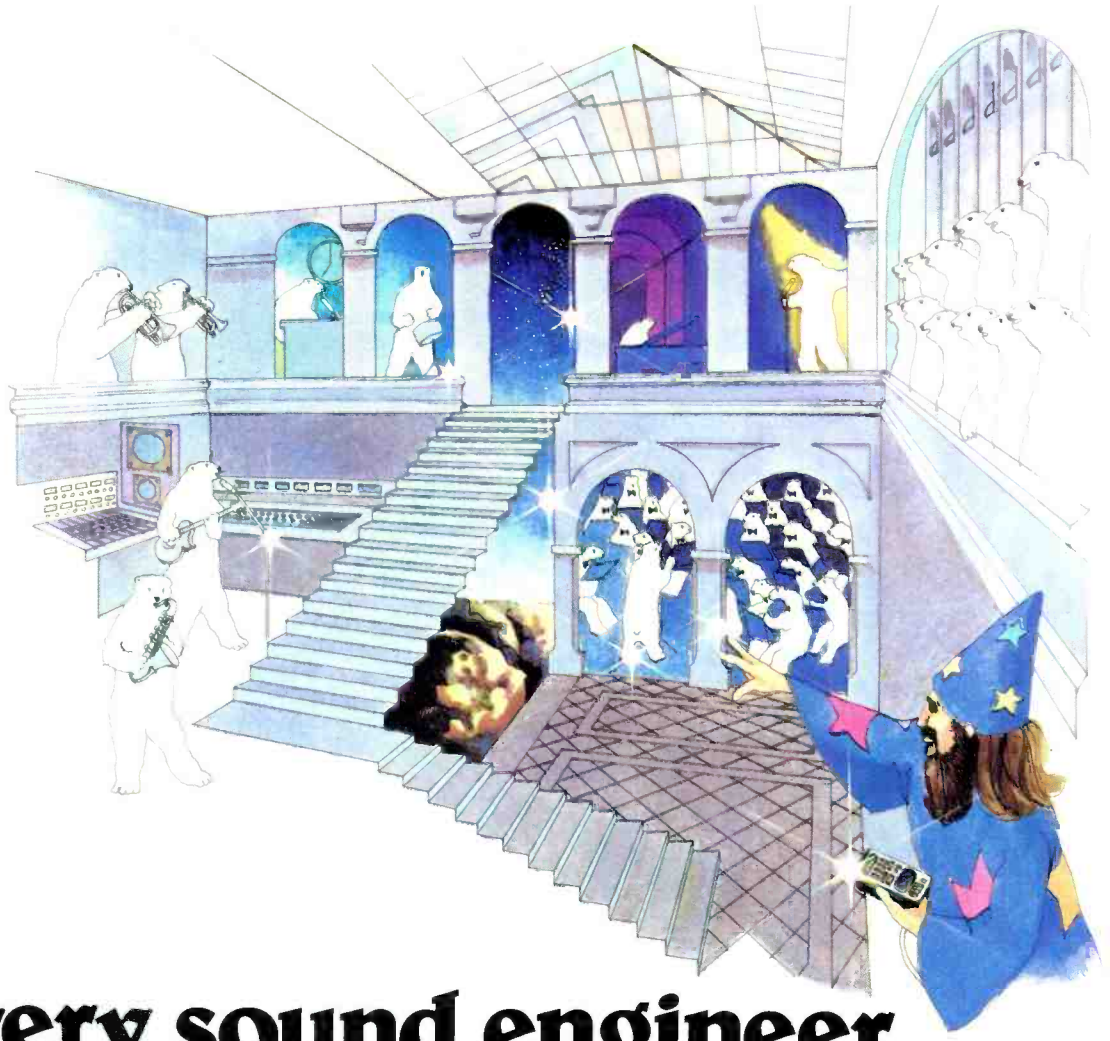
For additional information circle #4



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.

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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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For additional information circle #5

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No other studio mastering tape is more consistent.

No other mastering tape is more available, either. With Ampex Grand Master 456 you have the confidence of knowing we stock our tape inventory in the field. Close to you. So we're there when you need us.

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For additional information circle #6

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It also offers you a much greater range of features than the vast majority of synchronizers. Including an edit list capability of up to 200 edit points.

And it's the only synchronizer developed by both a professional audio/professional video manufacturer.

But the real reason for buying it is that it is upwardly compatible to the proposed SMPTE "Recommended Practices for Digitally Controlled Equipment." Which means the interfacing problems between video, audio and film equipment will be problems of the past. This Sony "Sync Master" synchronizer has a built-in distributed intelligence network that makes it able to talk to an entire universe of diverse machines developed by diverse manufacturers.

So before you invest in a synchronizer that just solves today's problems, perhaps you should first examine the one that will also solve tomorrow's.

SONY
Professional Audio.

For additional information circle #7

views letters neu

NOISE LEVEL MEASUREMENTS — A Reply

from: **Don Davis, president**
Synergistic Audio Concepts
San Juan Capistrano, CA

Mr. Buff misunderstands the intent of my previous letter [published in the February issue of *R-e/p* regarding Paul Buff's article, "Performing Meaningful Noise Measurements," — *Editor*]. It was not intended to teach (who can teach another?) but rather to offer him a "learning" opportunity. The "gain or loss" of a device is that value expressed in N-dB that occurs at the *output* of a system (i.e., a sound level meter at the listener's ears) when the device is "inserted" into the system. It is this value that allows "gain charts" to be constructed. Mr. Buff's acceptance of:

G = 20 Log (EOUT/EIN)

is a common failing among circuit designers, but is not a virtue to *system* designers.

Back in 1924, W.H. Martin of the Bell Telephone Laboratories wrote:

> "It should be noted particularly that *the change in output power* of the system is the *real measure* of the effect of any part of the circuit on the efficiency of the system that the ratio of power leaving any part to that entering it is not necessarily the measure of this effect... For example, a pure reactance placed in series between the transmitter and the line may change the power delivered to the line by the transmitter and, hence, the output of the receiver, the magnitude and direction of change being determined by the impedance relations at the point of insertion . . . The ratio of the power leaving the reactance to that entering it is, of course, unity, as no power is dissipated in a pure reactance . . . In other words, the transmission efficiency of any part of a circuit cannot be considered solely from the standpoint of the ratio of output to input power for that part, or the power dissipated in that part, but must be defined in terms of its effect on the ratio of output to input power for the whole system."

To envision what Mr. Buff is failing to account for, the complete equation for gain is:

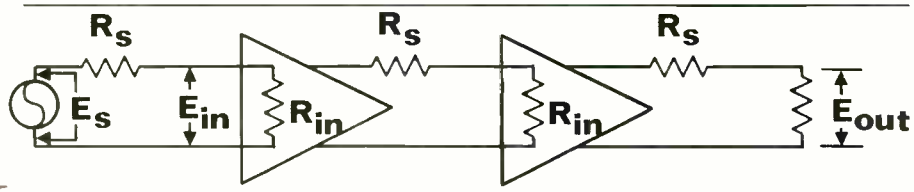
G = 20 Log (EOUT/EIN)
+ 20 Log [RIN/(RS
+ RIN)] + 10 Log (RS/RL) + 6.02 dB

Where: G is the insertion gain or loss,
EOUT the output voltage measured across the load,
EIN the input voltage measure across RIN,
RIN the input impedance,
RS the source impedance, and
RL the load impedance.

On rare occasion voltage amplification and gain are equal (i.e., the matched case).

Mr. Buff's attack on myself, the ancients, and good engineering practice in general smacks of "Ready, Fire, Aim." We suspect that it is Mr. Buff who is swimming against the tide. I am pleased to be lumped in with the ancients for whom I have the greatest respect, and whom I believe will be found to be still quite relevant and applicable to tomorrow's problems.

One final comment: When gains and losses are computed as I have described, then the *acoustic* level of a performer may be *directly added* to the EIA available input power level rating (i.e. sensitivity rating in AIP in dBm for 0 dB LP), and the electrical output power of the electronic devi
can be *directly*
tivity value for
30-foot, one-mi
obtain the acoustic level at the listener 30 feet in front of the loudspeaker. No such simple transition from acoustical to electrical to acoustic levels exists in Mr. Buff's world of voltage amplification.



THE GAIN OR LOSS OF A SYSTEM

There may be any number of devices having both gain and loss between input and output of a *system*. When *the gain or loss of each device* is correctly computed *they may be added together* and their sum will equal the total system gain or loss. The output level of a device in between the source device and the load device has its output level computed as an available input power (AIP) level to the following device:

AIP level in dBm = 10 Log [(Es)²/ 0.001×Rs] — 6.02 dB

DIGITAL/ANALOG COMPACT DISC

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

Thanks for the article on mastering for Compact Disc by Roger Nichols [February and April issues of *R-e/p*]. As a potential consumer of CDs, I'm all for sanity in high places, but it seems to me that it may be too much to ask, especially considering how the major labels often re-master analog disks.

I recently bought an audiophile edition (if I'm allowed to name names, it was the half-speed mastered version of Mike Oldfield's *Tubular Bells*, from CBS), and found that it had been cut from the *quadrophonic* (quadriphonic?) mix, though it had not been encoded for quad. I'm frightened of the Compact Disc version — there's no telling what it might be like!

HEADPHONE DISTRIBUTION SYSTEMS

from: **Alan Fierstein, president**
Acoustilog, Inc.
New York City

The article "Designing Headphone Distribution Systems," [by Rick Simon; April 1984 issue], makes the point that two-watt pots are needed with a 250-watt amp, because such an amp puts out 44.72 volts RMS. I point out that this presumes you are feeding a full-power constant tone to the musicians' headphones, which of course will not intentionally occur in actual use. Musicians are being fed *Music*, which has peaks 10-15 dB over the average. If the amp is just at the edge of clipping, the RMS voltage will be in the vicinity of 7 to 14 volts.

The peak value of a sinewave is higher than the RMS value by a fixed ratio. However, with music the voltage level "peaks", but not necessarily on a regular basis, and not necessarily dur-

. . . *continued overleaf* —

...TERS

...le of music. In fact, I am not ...use the term "cycle of music" ...e an oscilloscope histogram (a ...n frame of a 'scope picture) will ...ten show no discernable "cycles."

We can sometimes tolerate slight clipping (1-2 dB) in music, but when kept clean the pots would only need to handle 0.78 watt. It is good practice to overspec this to two watts. Interestingly, we arrive at the author's recommended value, but by a totally different route! If tone is accidentally fed into the system, an even higher power pot is needed, because the clipped wave has more heating power.

A more serious error is the author's recommendation of an audio taper pot. Such a pot is 20 dB down at midpoint. I feel it is logical to assume that musicians will set them at midpoint. When this happens, the amp could be putting out full power with music just clipping, but the sound would not be loud enough for many musicians. What frequently happens next in this situation is that the musicians complain to the engineer, who turns up the level at the console, and the amplifier is driven further into clipping. The all the pots start to burn up. Meanwhile, in response to complaints about the distortion, some poor assistant engineer puts a set of cans on his head and goes over to a cue box which happens to be turned up full and plugs in. Although this scenario may sound humorous, permanently damaging someone's hearing is tragic.

One solution to this is to use a linear pot, which is only 6 dB down at midpoint (a little lower when a typical headphone is plugged in). They are also cheaper and more widely available. Also, the control room should have a calibrated meter on its cue system, which I find only rarely in my clientele of studios. Finally, if your cue system has pots, get in the habit of turning them up full before sessions.

SOUNDMAN'S UPDATE

from: **Mark A. Holman**
Pulsar Laboratories, Inc.
Mogador, Ohio

First of all, I would like to congratulate you on an excellent magazine. I really enjoy and have benefitted from your articles regarding sound companies and audio-for-video, as well as studio equipment and procedures. The reader response we have received by advertising in *R-e/p* has been very good, which is why we intend to continue advertising in the years to come.

There are two things I want to bring to your attention regarding the April issue of *R-e/p*. The write-up by Andy Chappel, "Soundman's Notes From The Road", on page 80 neglected to state two pieces of information. Regarding the review of the Cleveland Agora, the house console is a Pulsar 32×8. Hood Industries' phone number is: (216) 431-4663. If you would make a note of this in your next issue we would greatly appreciate it.

EXPOSING AUDIO MYTHOLOGY

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

by John Roberts

For those of you who haven't noticed, it would appear that for some time now, quite a few of us in the recording industry have been spelling a word wrong, or at least differently. The rest of the scientific establishment spells the word "bus" with a single "s", while we choose to spell it "buss." A quick peek in the dictionary indicates that "buss" is something you do with a member of the opposite sex (look it up), whereas "bus" is the word we really want.

My question for you is where did the spelling of "buss" come from? I recall the startled look I got from the editor of a British technical magazine, when in the course of a conversation about words that we spell differently I mentioned bus(s). It turns out that the proper British spelling is also bus. I spelled it wrong in blissful ignorance until noticing in a technical journal that computers used buses, sort of like our busses but without the buzzes (sorry about that). [*R-e/p*'s house style is singular "bus" but plural "busses" — *Editor.*]

This topic may take the prize for the least important Audio Mythology topic I've ever discussed but, who knows, there may be an interesting story here. I would appreciate early examples of either usage, but especially "buss." I have found "bus bar" mentioned in a studio wiring diagram dating from the Thirties. Note: IEEE S-100 spells it "bus."

Slew Rate

I feel that slew rate is grossly overrated as an indicator of sonic performance — unlike distortion or frequency response, parameters that can predict colorations which affect the quality of reproduction accuracy. Slew rate is more like a power output specification: you should always have enough to pass the signal with sufficient headroom to avoid clipping (slew rate limiting); beyond that, however, the benefits are questionable.

The only catch that I can see to this simplistic analysis is the fact that not *all* amplifiers slew limit in the same way — some become non-linear (distorted) well before limiting, sort of like tape saturation but nastier sounding. It does not help that most manufacturers specify slew rate from op-amp to power amp with the input stage over-driven. Which is not much different than specifying output power while letting the signal clip. You'd be surprised how much power you can get out of your power amp if you let it turn the sinewaves into squarewaves!

We long ago learned that for a power specification to be meaningful, we must also know the distortion produced at that output power. An even better example is recording tape. To properly spec a tape's signal-handling capability, you usually are quoted a given level and then a distortion figure for that level. Most amplifiers slew-rate specifications are made at what must be 10 to 30% distortion (a guess). There is little incentive to specify slew rate any other way because it is perceived by the marketplace that "more is better."

It's Not How Fast You Make It, But How You Make It Fast

If every amplifier used similar topology, you might be able to compare slew rates and infer that Amplifier A might sound better than Amplifier B because it has a slightly faster slew rate. However, it is possible for a slower amplifer to have a cleaner top-end response.

While it is beyond the scope of this column to discuss the kind of circuit techniques available to power amplifier designers to optimize high-frequency performance, let me state that any professional power amp of recent design will, more than likely, have adequate slew rate — and even if it didn't you probably wouldn't be able to tell from the spec.

It is difficult to characterize exactly what slew-overload sounds like because different topologies will behave differently. In fact, it is possible (see reference #1) to design an amplifier that cannot be slew limited by any signal that wouldn't also cause amplitude clipping. Very fast signals are effectively low-pass filtered. In theory such a topology could be used to design a relatively slow, yet impossible-to-slew-limit amplifier. But would it sound good? Yes and no. It would sound better than a slow amplifier that didn't avoid slew limiting, but

. . . continued on page 19 —

Do you know that 9 out of 10 of the most successful studio owners in Mid-America buy from Milam Audio?

I'm really not interested in anything but price ... What's your bottom line?

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Finally, somebody took digital recording from here

COMPUSONICS INTRODUCES "TRUE" DIGITAL SOUND. FROM MIKE TO MASTER. IT'S WHAT DIGITAL SHOULD HAVE BEEN ALL ALONG.

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Because the majority of digital masters were, in fact, produced on analog tape equipment and then merely converted into a digital format.

But a new company called CompuSonics has developed a multitrack digital audio mixer/recorder that allows you to produce digital recordings from mike to master. Without a single analog step in-between.

Introducing the CompuSonics DSP-2000. With it, an engineer records from start to finish with digital sound.

CompuSonics

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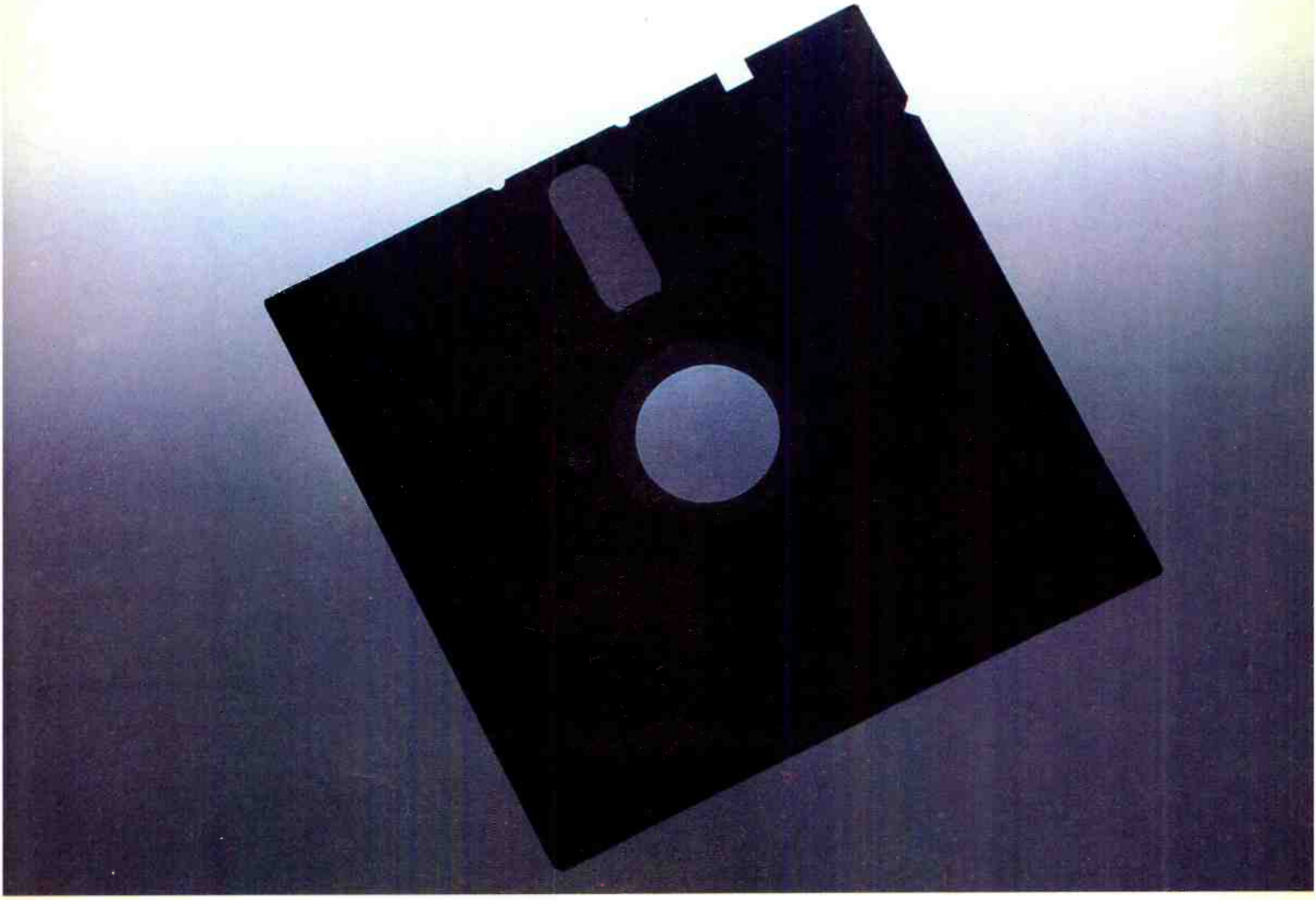

True digital sound.

Without analog compromise. Which means the elimination of signal loss on dubbings and re-mixes. High end loss. Tape hiss. Degradation. And background noise.

So from now on, the only sound you'll hear is the one you intended to put down.

And because the DSP-2000 is a multi-processor computer it offers benefits conventional equipment can't. For instance, with the computer you have instant access to file positions without rewinding miles of tape. Plus the advantage of digital signal processing. Allowing you to control your sound.

To shape it. Play with it. Store it. Recall it. Edit it. In your choice of formats.

But all of this is just the beginning of what the DSP-2000 can do. Now that you've heard about it, hear it. Call your CompuSonics sales representative at 1-800-223-1821 for more information. He'll take you into the digital sound of the future. And that's true.

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For additional information circle #9

Three hundred years ago, it took the hand of a master craftsman to create a concert violin. It also took commitment—a commitment to producing the highest quality, purest possible sound.

Today, at the new Harrison, we still follow the craftsman's tradition—and we share that unwavering commitment. We realize that, even today, there is no substitute for the enduring quality and value of precision craftsmanship. That's why we take the time to listen to your ideas and needs, and why we use that input to build every Harrison console with the same painstaking care that the master violinmaker devoted to his craft.

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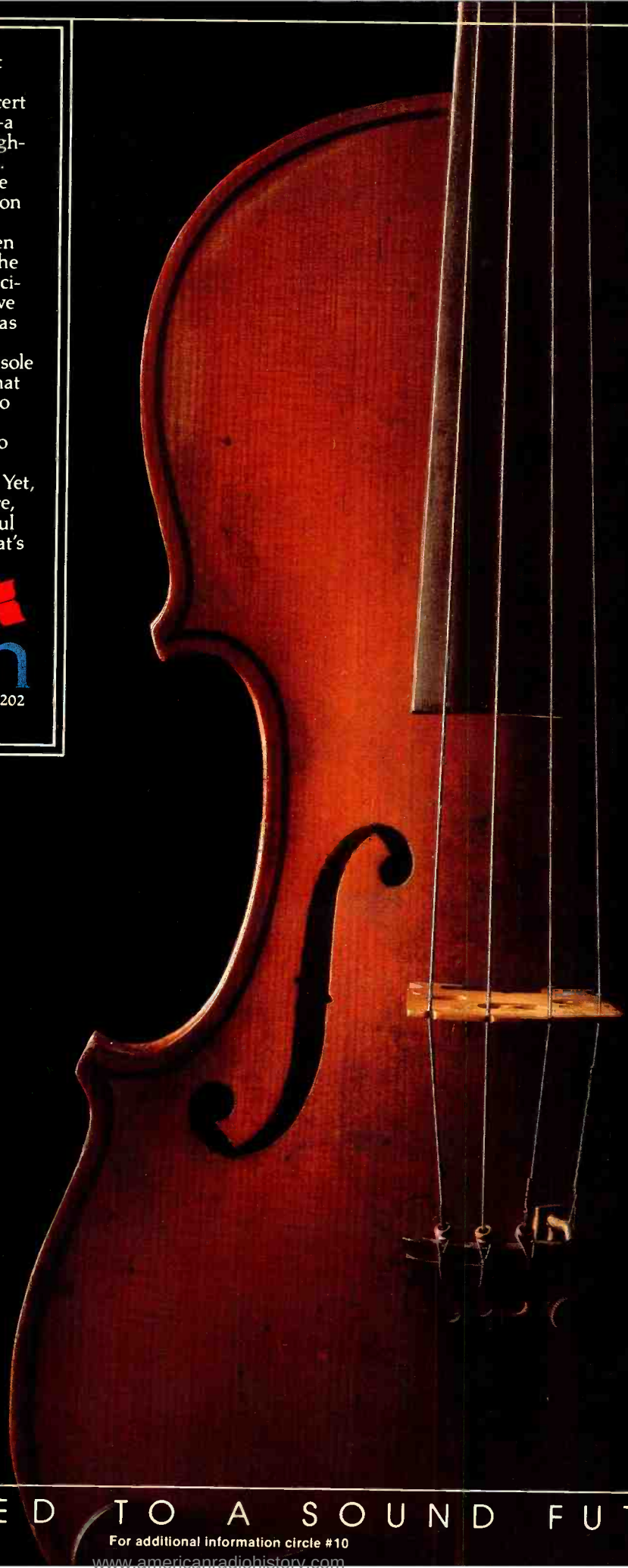

COMMITTED TO A SOUND FUTURE

For additional information circle #10

— *continued from page 14 . . .*

would be subtly inferior to a wider bandwidth design.

While I have no doubt that amplitude accuracy (frequency response) is still the most significant determinant of sound quality, the ear/brain/listener is also sensitive to temporal errors. I won't be brave enough to guess how much time distortion is problematic, but it appears that the gross errors caused by the multiple cascaded filter stages used in early consumer Compact Disc players were at least partly responsible for the less-than-sweet high end reported by more critical listeners.

Op-Amp Selection

When it comes to selecting op-amps, it's a little easier because the most popular general-purpose devices are either faster than we need, which is okay, or horribly slow and fairly easy to eliminate. The slow op amps (741, 324, etc.) generally exhibits saturated slew rate of ±0.5 volts per microsecond. This is too close to the actual slew rate of musical signals, leaving inadequate headroom for safety from overload, nor any margin for non-linear operation prior to slew rate limiting. I have seen these op-amps used successfully (although some would debate how successfully) in lower-cost mixers; typical "trickery" includes dropping the signal level they see, and, in lower-cost processing gear, by carefully using the slow op-amps in circuits that see slow signals (like a crossover's low-frequency outputs). In some cases the use of the op-amps can explain the less-than-crystaline sound of some middle-aged IC gear. (Tube equipment, by the way, is quite fast, and generally not bothered by high slewing signals.)

The second group of general-purpose op-amps (the fast ones) are dominated by the BIFETS. The FET input amps can usually slew at ±13 volts per microsecond (TL070, LM351, LM347, etc), with a few tricked-up Bipolars in the bunch, most notably the NE5534 at ±10 to ±12 volts per microsecond slew rate. While, at first glance, these amplifiers appear to be an order of magnitude faster than we need, sensible headroom and linearity margins make them just right.

For the Money-is-No-Object crowd, there are hybrids (discrete) amplifiers that go faster, and, in some circuit applications, actually sound better than their off-the-shelf bretheren — although, more likely than not, this is the case not just because they're faster. There is more to circuit design than slew rate, but that will have to wait for another column. (See also my article in the April 1984 issue of *R-e/p* for a discussion on Negative Feedback.)

TIM, SID, *et al*

I couldn't write about slew rate without mentioning TIM (Transient Intermodulation Distortion) and SID (Slew-Induced Distortion). TIM is just a fancy name for slew-rate limiting, and SID describes the rather vague region between linear operation and slew-rate limiting. Once again, to use the tape saturation analogy, non-linearities prior to slew-rate limiting can cause audible distortions similar to tape distortion before it clips.

While it should be a simple matter to characterize an amplifier's slew rate performance by simply increasing the frequency/amplitude of a sinewave until you reach 3% THD, the nature of a THD analysis is to measure the disortion products that are higher harmonics of the test signal. Unfortunately, the mechanism that caused the slew-rate limiting will attenuate the higher harmonics, making such measurement unreliable.

Champions of THD and SID proposed a dedicated TIM test that consisted of mixing a sinewave with a squarewave, and looking for slew-related distortions on the sinewave. Although this technique was effective at pointing out slew limiting, the test never met widespread approval.

However, it seems that the technique's proponents were successful at raising the general awareness of high-frequency linearity, and Two-Tone Intermodulation Distortion tests have become more popular. A two-tone test using 19 and 20 kHz sinewaves will stress an amplifier the same as a 39 kHz single tone, and will generate an easy-to-measure distortion product down at 1 kHz. This test appears to correlate well with actual performance, and overcomes the limitations of using out-of-band test signals. No one will argue that a distortion product at 1 kHz isn't important, but try to sell the significance of THD at 39 kHz!

In conclusion, if you're shopping power amps, don't attach too much importance to slew rate specs as they

— *concluded on page 22 . . .*

"DESIGNING HEADPHONE DISTRIBUTION SYSTEMS" — A Correction

A couple of typesetting errors in Rick Simon's article published in the April issue of *R-e/p* may have caused a certain degree of confusion. The three equations in the center column of page 170 should have read as follows:

1) $V_L/R_L = I_L$
2) $P_{ave} = (V_L)^2/R_L$
3) $P_{ave} = I^2 \times R_L$

Unfortunately, the power-two expression was omitted in these formulae, and also in subsequent mentions in the article's text. In addition, the regulated power supply specified to drive low-impedance headphones should have read ±5 volts (and not +5 volts), while the power supply specified for driving 600-ohm cans should have been ±35 volts (and not +35 volts).

Our apologies to all concerned for any confusion that may have resulted from these errors — *Editor.*

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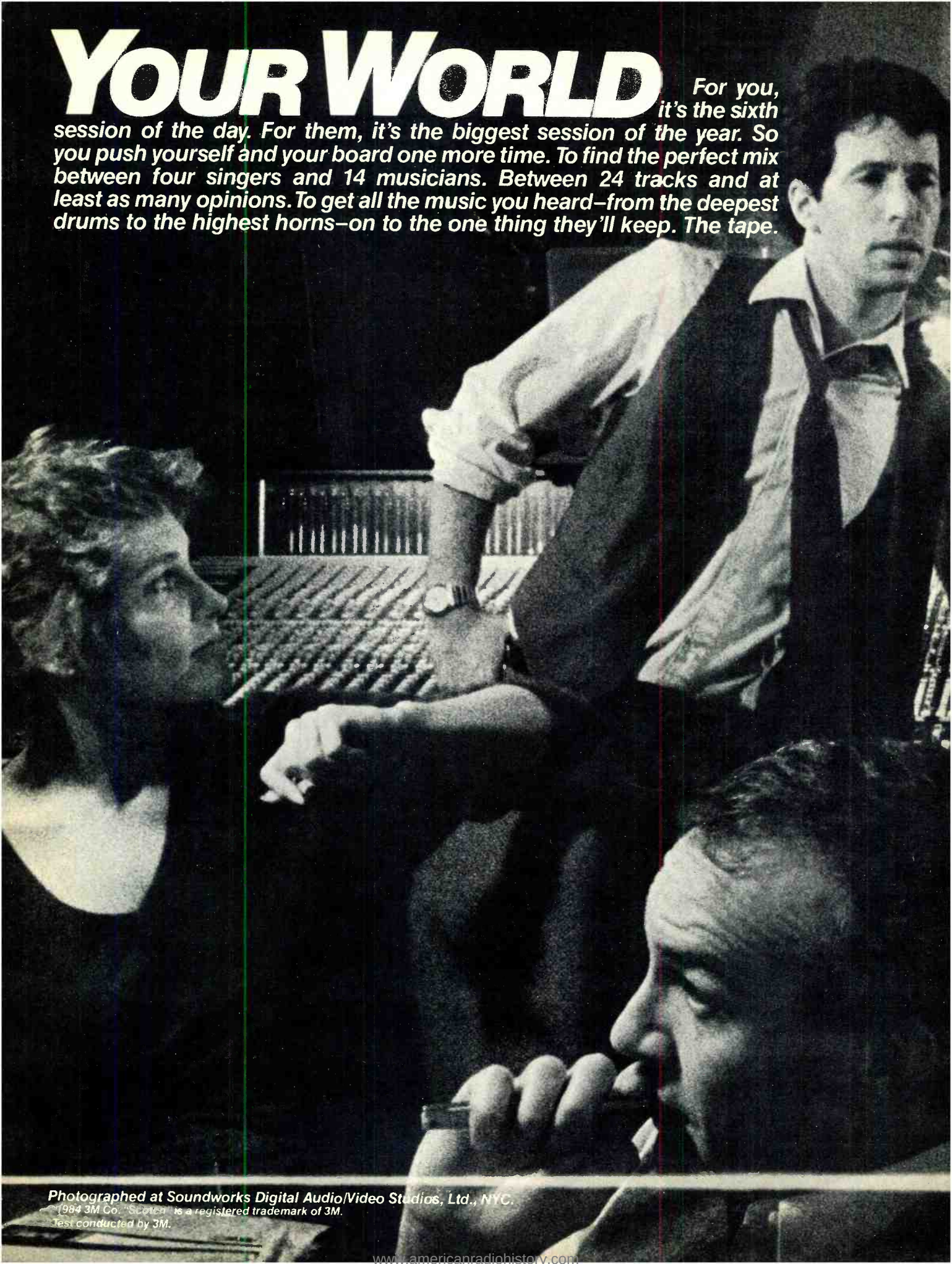

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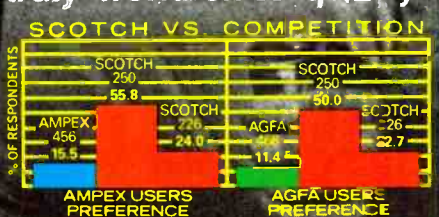

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— *continued from page 19 . . .*

are not always comparable from amp to amp. If you're designing a circuit, spring for the faster op-amps; they really don't cost that much more (60¢ to $2 in small quantities).

REFERENCE:

1. W. Marshall Leach, "Input Stage Design Criterion for the Suppression of Dynamic Distortions," *J. Audio Eng. Soc.*; Vol. 29, No. 4; pp. 249-251 (1981).

news

BILL ROBINSON — A TRIBUTE

William L. (Bill) Robinson died on April 19 at the age of 67. His constant pursuit of technical excellence for over five decades leaves behind a legacy in the recording industry that will be felt for generations to come. He was considered by many to be a master in the art of sound, and pioneered many innovations in the areas of stereo recording, disk recording, and multitrack recording, as well as his innovative designs for echo chambers, isolation chambers and sound baffling. Bill was perhaps best known for his valuable contributions at Capitol Records and Sunset Sound Recorders, in addition to his many contributions to the AES, and also as an independent consultant. He received the Society's highest award not once, but twice, and was also bestowed an honorary membership of the AES in 1981 for technical leadership and pioneering in the fields of recording and broadcast operations.

At age 16 he was made chief engineer at a Los Angeles radio station; at the age of 18 he moved on to New York City to join WABC and then to WBBN in Chicago. Returning to California, Bill freelanced as an engineer for many stations until 1937, when he joined the Mutual Broadcasting System as a field engineer for remote broadcasts.

After the war, Bill started working in Los Angeles recording studios, eventually becoming Director of Recording at Capitol Records in 1950; a position that he held until his retirement in 1969. Bill's contributions during these two decades at Capitol resulted in a string of

. . . *continued on page 36* —

INDUSTRY INVENTIVENESS IN THE EIGHTIES

Fin Johnston's Cost-Effective Music Production for Commercials and Jingles

by James Riordan

While record sales seem to be on the rise, most music industry observers are wisely sticking to their money saving ways. This writer is optimistic about the total recovery of our industry, but we would be showing prudence to regard last year's record sales with a grain of salt. (Just imagine what they would be if one person — Michael Jackson for instance — had decided to postpone recording until next year.)

One positive sign, which clearly illustrates the adage of "Necessity the Mother of Invention," is the tendency to use crea-

tivity and human resource as a substitute for big spending. Everyone is realizing that, in most cases, the same things can be done as well for less money. More and more people are figuring out new ways to save people money yet still provide them with a quality product or service; Fin Johnston, the subject of this month's column, is a good example.

Providing music for radio and televison commercials is big business, and there are countless jingle companies and music houses handling such work across the country. However, the well known drawback to using prepackaged jingles, sound effects, or music is that the advertiser must sacrifice individuality and customized sound for a substantial savings — hundreds of banks, for example, have the same jingle with just the name substituted. Although the jingle company agrees not to sell the same jingle to competing institutions in the same area, there is still little room for a personalized touch.

Major television and radio advertisers hire their own composers, musicians, and studios to give them exactly what they want. Many top music houses offer their own recording studios, staff composers, lyricists, producers, and musicians to meet this demand — they also have a substantial overhead which, of course, must be passed along to the client. This may explain why many small or regional advertisers can't afford customized music and must use prepackaged material. And that was exactly the need that Fin Johnston saw, and set out to meet.

A composer and multikeyboardist, Johnston moved to California five years ago to pursue a recording career. He and his brother Kevin joined up with bassist Tom Fowler of Frank Zappa's Mothers of Invention and guitarist Tony Creed from the Surfpunks. Playing the bar and lounge circuit in L.A. is not exactly a lucrative venture, as most musicians soon discover. An excellent jazz pianist, Johnston supplemented his income with an occassional studio date. One afternoon he was asked by his landlord if he would be interested in doing the music for a low-budget television commercial.

"I've been playing keyboards for about 18 years," he explains. "I had always been interested in writing jingles, but I never pursued it. My landlord at the time was a special effects man for films, and he recommended me for a commercial on a car dealer in New Mexico. It turned out pretty good and the word got around.

"A lot of ad agency and commercial production companies are looking for people who can do a good job writing and recording the music within a reasonable budget. A couple of people really helped me out by referring me. One of the was Cheri Hunter, who is an excellent editor, and another is Arnie Lerner of Lerner Film & Tape, who I've worked with quite a few times."

Despite his desire to write and record his own songs, Johnston found himself spending more and more time creating and recording music for commercials. Over the next few years, he realized the tremendous potential of this market, and began to zero in on it. According to Johnston, there are several advantages he can provide. First, he is a songwriter, and can totally customize the music to fit the visual image the client is trying to convey.

"Often I'm dealing with people that have very little, if any, musical knowledge," he says. "They're totally relying on me to do it right. Sometimes they don't even have an idea what it's going to sound like in the end. I'll get a call or meet with the client, and they'll explain what they want and read the script. Occasionally, I'll see the video on a Movieola, and sometimes they will give me a color work chart which explains in colors where different things come in. The top line has how many frames, and the line below has how many seconds have passed.

"Generally, everything's being done at once and I'll have to work from the script. When I get the first call they may be just starting to shoot on location. By the time I come in to play my demo a few days later, they're already beginning to edit. The schedule is usually very tight, and they're depending on me to have it right."

Another advantage Johnston says he can offer clients is that his talents include vocalist and multi-instrumentalist playing keybords, guitar, bass and synthesizers, including electronic drums.

"On the last one, I layed down electric drums first using a Doctor Rhythm Graphic running it direct on one track. Next I played an electric bass track, and then put down the synthesizer doing chords. I run everything direct unless we use an acoustic piano. Sometimes I'll add a guitar track or percussion. Then I may have two or three tracks of effects depending on the commercial."

Johnston has his own fully equipped

. . . continued overleaf —

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demo studio that enables him to present the client with a complete picture before going to the expense of a commercial studio. Included are several instruments, the most essential of which is his synthesizer.

"There is so much demand for good sound effects that you *have* to have a decent synthesizer. It should be able to save the programs, because logging the dial settings isn't really accurate enough to be sure you get the same sound. It should also have at least a couple of oscillators.

"I usually put two or three ideas down for the client to hear that I feel are along the lines of what they are looking for. Sometimes, we'll just record the master note for note like the demo. Other times, the client will completely change direction, realizing that his original idea won't work. That makes more work for me, but ensures the client is satisfied. By demoing everything first I can save them a *lot* of money in experimental studio time."

One of the keys to Johnston's success, he considers, is his ability to create attention getting synthesizer effects that are customized to the individual client and commercial.

"I have real good communication with the clients," he offers. "I don't mind working to help them get a specific sound. A lot of times they can't express it musically; they'll make a sound with their mouth and want me to create something close to that. I'll start out with something basic, and they'll want it fatter on the bottom or the tail to last a little longer. They could search forever in one of those jingle houses and not come up with it. I can create it with them and refine it to get exactly what they want."

Another important advantage to clients working with Johnston is that they only have to deal with one person for all the music and sound effects, which eliminates the hassle of tryng to communicate effectively with several different people involved in composing, arranging, producing, performing, and recording a music and effects track.

"There are so many people involved in creating a commercial that simplifying things is a *big* help. They could go to a jingle house and pick out sound effects. However, not only does it usually cost more, they have to spend time finding each effect, getting it transferred to a quarter-inch tape, worry about the music, who's playing on it, how much is the studio going to cost, what if they decide to add something on the spot, and so on."

The most important consideration for most clients is that Johnston can handle all of this for a fraction of what it costs to hire a music house or separate individuals. "I can cut corners to make the budget work. The more money they have, the better studio we can go into. If the budget is good, we can use a studio with digital readout and a video monitor to work by. Sometimes the budget isn't big enough to allow for this kind of thing, and I have to compensate. Sometimes the schedule is such that I will have to do the music without a video, because the work copy is not yet finished or available.

"It's nice to have [the video] to work with, but if the piece is already written, and you're sure that the video will stick *exactly* to the script, you can get by without it. If there is a lot of specific sound effects to the video that have to match perfectly, it's a lot harder without the video because it has to be timed so accurately. If it's just wall-to-wall music from beginning to end with no particular drop-out or builds in certain spots, there's no reason to go to the extra expense of working with the video.

— Fin Johnston —

"The kind of equipment you're working with in the studio is one of a lot of things that vary with the budget. I'm willing to work around this more than most people who do my kind of work."

Johnston is also interested in the video side of commercials, and plans on expanding his knowledge in this area. "I've gotten in on the video aspect of it a few times, and given my input. I've only had a taste of it, but I try to learn as much as I can because it relates so closely to my work."

The recording process for a low-budget television commercial is relatively consistent, as Johnston explains: "The voice-over is usually done last, although sometimes there will be a work vocal to use for reference. Most of the time I have to work from the script, but if there is a video it's in the form of a 35mm print; sometimes we'll make a videotape copy to work by.

"We master to two-track, but it's usually a mono mix unless it's going to be used for radio. We record on one-inch/eight-track tape most of the time, and sometimes we'll do a submix. We'll always leave the effects on separate tracks so we can have control over them.

"I might combine a lead synthesizer that is playing the melody or rhythms with a bass, and maybe some kind of a rhythm track into a submix. Any effects like explosions, or lasers, or wind, for example, would be left on separate tracks. We might decide to pull an effect out completely and not use it at all, and sometimes we'll do a lot of mixes with the effects at different levels. It really depends on the project, but usually we'll lock into what we want and maybe do two master mixes."

The recording quality for a regional television commercial that may only be shown in Oklahoma City must be as good as a national spot, but it's not as demanding as most music recording sessions, Johnston considers.

"The quality of the recording must be very good, but the chances are that it's

only going to be heard out of a five-inch television speaker. I use a Fostex 250 [four-track] cassette recorder for demos and, in a few instances, we actually wound up using some of the effects right from these tapes. If it was only going to be four or six tracks it could be done on a four-track machine —especially if the sound effects are going to be laid in loose.

"One way to put in the sound effects is to have them timed exactly right in real time with the tape rolling. Let's say that at 10.32 seconds this one sound comes in, and it's got to be out by 10.45 seconds. You can roll your tape up to that spot, cue the engineer, and hit it. It's hard to get it right that way, and you usually have to do it a few times.

"The other way is to do it loose on a quarter-inch tape, and they'll take in all the sound effects and lay them in just right while they're watching it in the editing room. There are drawbacks to that too, in that it's usually done at the final editing stage when I'm not there. I've heard a couple of things I did that were off a half second or so during the audio-sweetening process. I like to be there whenever possible to make *sure* it goes right."

Anyone who has ever worked in the field has something to say about the lack of emphasis given to the music in commercials, but Johnston is willing to work around this as well. "Sometimes you will hear a commercial where half of a measure is missing, and it sounds terrible to anyone with a musical ear. Usually this is because they had to chop the music to make it fit; they might leave a bar or two out or change to another section of the piece that is in an entirely different key. It's musically incorrect, and it *really* devalues the worth of the music.

"If there's no definite tempo you can just about bring things in whenever you need to, but when you're going in time and the first effect comes in on time, but the second one *doesn't* it can be very difficult to get it musically right and visually workable. You have to mess with the tempo a bit so that you come out where you need to musically, and not end in the middle of a phrase. Sometimes, the music runs all 30 seconds of a 30-second spot, but other times it has to be out by 28½ or 29 seconds. The timing has to be *just* right." ■■■

VISUAL MUSIC SCENE

Audio Post Production for Videotaped and Filmed Concerts

by Adrian Zarin

In the post-MTV world of the promotional video clip, it's easy to lose sight of the humble concert video. But, as some may recall, videotapes of live rock performances enjoyed an early-Seventies vogue with the proliferation of late-night network shows such as *In Concert* and *Don Kirshner's Rock Concert*. These shows easily eclipsed lip-synced daytime programs of the *American Bandstand*-mold. At the same time, they filled a void that had been left vacant since the demise of mid-Sixties rock TV greats like *Shindig* and *Hullabaloo*.

Concert video was, in turn, eclipsed by the promotional "concept" clip. While no longer the center of attention, however, the concert format continues to thrive on both cable and the networks. Not surprisingly, today's concert videos have been given a creative shot in the arm by their counterparts in the "conceptual" arena. The use of more inventive camera angles in recent concert videos, more resourceful cutting, and freer use of post-production special effects are all factors that betray the influence of the video clip.

Audiowise, concert videos afford much more room for creativity at the technical level. Rather than existing as a pre-recorded, finished studio product, to which pictures are then added, the audio portion of the program assumes equal status with the picture. It is a living, breathing entity to be captured in its essence right on the spot — along with the picture — then distilled and enhanced in post production. Naturally, all the techniques of mobile audio recording come into play. But, as an added challenge, the audio engineer's resources and procedures must be carefully coordinated with the visual aspects of the program. Audio and video equipment needs to be meticulously synchronized, and open, flexible lines of communication must be maintained between audio and video personnel.

As audio producer and associate producer for NBC's recent *Rock and Roll Tonight* series, Patrick Griffith of PG Productions was in many ways an heir to the mid-Seventies late-night concert shows.

"We more or less got carte blanche from NBC," recounts Griffith, whose experience also extends to cable concert specials with The Rolling Stones and Hall & Oates, along with countless radio concert programs. "It was one of the most advanced projects we got to do. *Rock and Roll Tonight* was more or less what the old *In Concert* tried to be, but with 1980's technology. It involved applying state-of-the-art video practices to a live situation. The point was to bring

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the audience what was going on at the venue, *not* to create a new art form."

Taped at Perkins Palace in Pasadena, California, using seven or eight video cameras, *Rock and Roll Tonight* employed one of the API/3M-equipped Record Plant mobile trucks to record the program's audio portion. According to Griffith, the 24-track format was usually sufficient for recording the show.

"We laid timecode and video synchronization on tracks 23 and 24, and the audience tracks went on 21 and 22," he reports. "The remaining tracks would be band information — vocals, instruments, whatever. We always made a rough cut of the picture — before the actual video was mixed, if you will — with the same timecode numbers burned into it. That would go back to the Record Plant, along with the 24-track audio tapes, for sweetening and overdubs. The main equipment we used there was Sony 800-Series [¾-inch U-Matic] video decks, [Audio Kinetics] Q.Lock synchronizers and Studer 24-tracks. Of course, we'd sync the audio tape up to the Sony machines using the timecode. The [60/59.94 Hz] sync tone was really just there as a backup. We'd compare the burned-in timecode numbers on the video screen with the readouts from the Q.Lock and the Sony. If everything matched up, we knew we were doing okay."

Maintaining Audio-Video Sync

Audio and video tape synchronization is apparently less of a problem than one might assume in preparing concert videos. As the current director of operations and long-time staffer at Hollywood's Complete Post, Andy Zall has handled post production on a generous share of rock concert programs. He has had ample opportunity to observe what *can* go wrong with synchronization.

"We don't have many problems with syncing up any more," he notes. "Not with the new synchronizers that we've had in the last three or four years. They are so good that, if the code is good on the multitrack and on the videotape, we have no problems.

"We get into trouble when the tapes have gone into recording studios where the staff will transfer the code, but not lock the machine. In other words, they will just synchronize to the 60 Hz reference tone, and not to the timecode. But in the past three or four years this sort of thing has become less frequent, too. Many recording studios have moved over to video and added timecode to their own facilities. So record people have been gradually educated as to how to deal with timecode."

Even if there is a mishap, according to Zall, "you can save almost anything — depending on the problem. The main drawback is having a tape that is not locked to the timecode, and the audio is drifting as a result. If it is drifting at a constant rate, you can use a tape speed override to correct it. In other words, you can change the playback rate of the tape machine by a variable amount. If the audio is drifting at a rate that's *not* constant, then you have trouble. You may have to go back and remix or make a new master.

"As for other problems, if the timecode has been recorded or dubbed poorly on the tape, for example, you can either jam sync the timecode, or relay it on the tape again. As a rule, there's always something you can do."

Cast and crew at the Arlington Theatre, Santa Barbara, for the video taping and multitrack recording of an HBO concert special featuring Linda Ronstadt.

Post-Production Creativity

One of the differences between post production on a concert video, as opposed to a promotional clip, is that it affords the audio engineer much more of an opportunity to work directly with the artists and/or their record producer. During post production on a promo clip, the artist's presence is often superfluous — and even a downright hindrance, according to some video people. The artist can play a vital role, however, in concert video post production.

"With *Rock and Roll Tonight*, we taped three nights a week, out of which we would get two edited shows," Pat Griffith recalls. "In some cases, we had as many as 10 bands [to record] in three nights. So you would really *need* the bands in post production to tell you what was going on."

Since many concert-video productions involve taping the same show on several different nights, artists and producers provide valuable input by selecting the best performances — from a musical point of view — of each song. Of course, the video producer and director will be selecting the best material from a visual point of view, which may or may not coincide with the musical material the artists have selected. Careful coordination of efforts is required in order to arrive at a successful compromise between musical and visual values.

As Andy Zall explains, this coordination of efforts can become very involved, which was the case with a Fleetwood Mac concert he edited for HBO last year. "The band's record producer and the video producer decided on which parts of which songs worked better together," he recalls. "We went back and forth making picture edits to match the sound edits, and sound edits to match the picture edits. The record producer brought the audio tracks to a recording studio and mixed them. A lot of times, he would combine the best parts of each song. He might go from the first night to the second night to get a better guitar solo, and then switch back to the guitar track from the first night — just slipping the solo over the backing tracks. It was all electronic editing; he didn't actually razorblade any of the original tracks.

"At the same time, we prepared a cassette of the picture we wanted to use. Some of our video edits were made because the music producer had told us, 'I want to make audio edits here.' We assembled our own video and audio pieces, frequently exchanging cassettes and making adjustments. We used the same timecode, so it all matched up all the way down the line."

Zall goes on to say, however, that the Fleetwood Mac situation was not a typical one, because of the time frame involved. "Usually there's not that much time," he explains. "We had about four months with that project; normally, the audio mix and the video editing are done simultaneously at different facilities. That's how it was on a Pat Benatar concert I did for HBO, for example. On each song, while the record producer was mixing, we were editing. The music producer and the video producer would communicate by phone, comparing notes and working out compromises."

Correcting a Musical Performance

Overdubs are another aspect of post-

. . . continued on page 33 —

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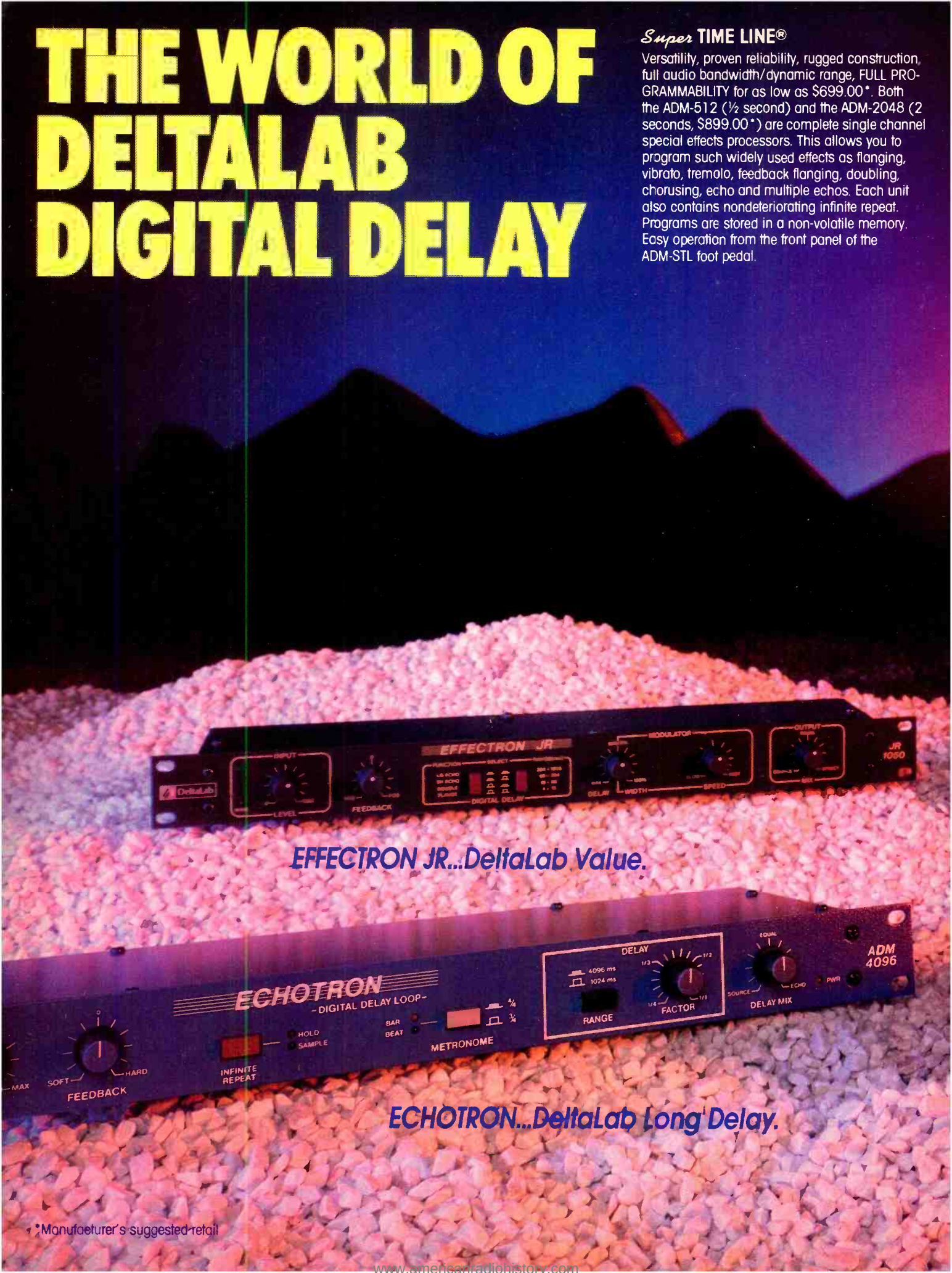

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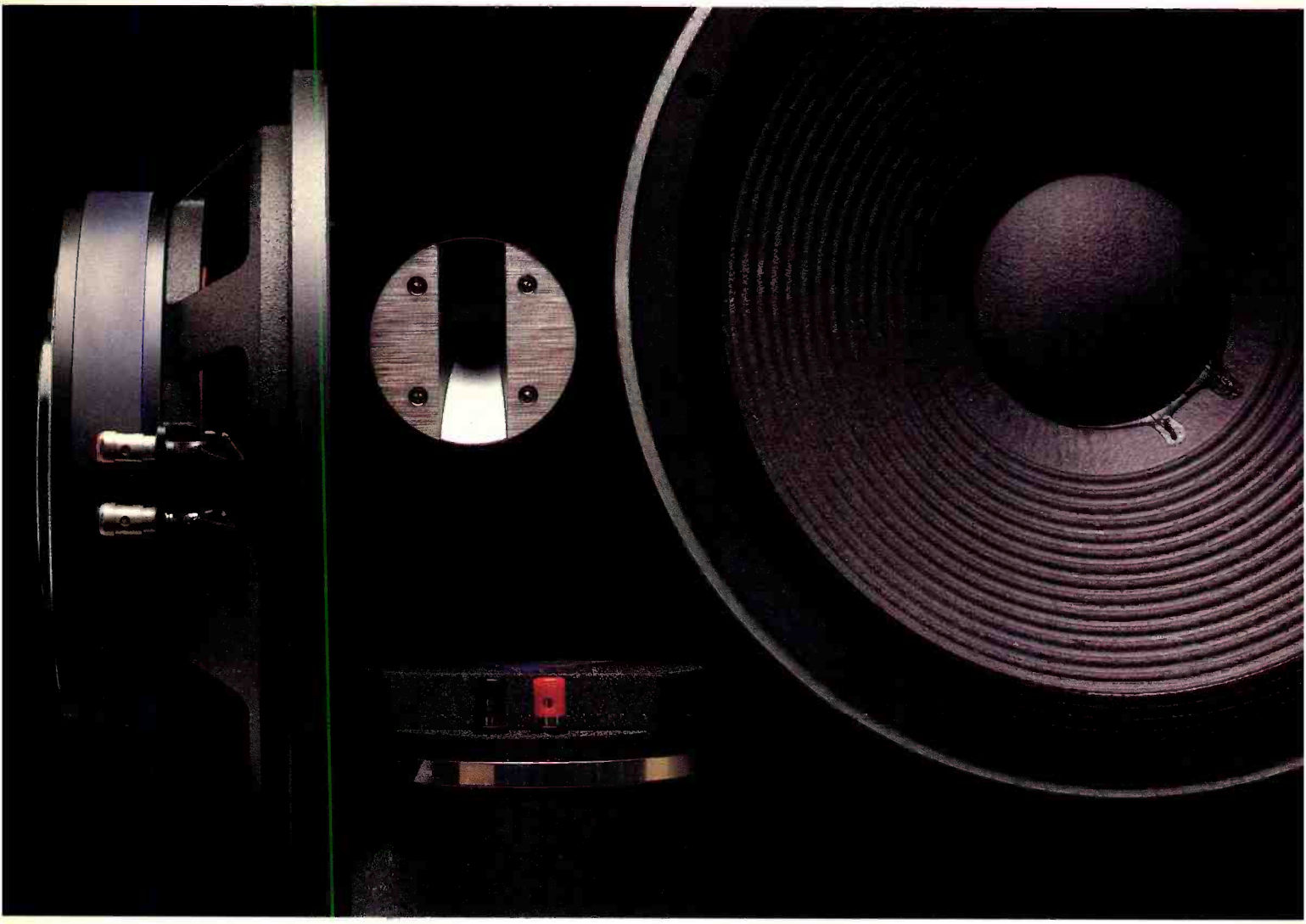

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— *continued from page 28 . . .*

production, an area in which concert videos overlap with record work. The end results differ, of course, since the goal of overdubbing in concert videos is merely to correct and enhance a performance as unobtrusively as possible. Most of the work focuses on fixing vocals and improving instrumental-solos.

Brad Gilderman is the audio engineer in charge of mixing and overdubs on *Rock Palace*, the show that has taken up the time slot formerly occupied by *Rock and Roll Tonight* on NBC. A large part of the job involves working on overdubs with *Rock Palace* guests after the show has been taped at the Palace, a Hollywood nightspot. Overdubs and mixing are handled at Cherokee Recording in Hollywood.

"The bands are not allowed to go *too* crazy with overdubs," says Gilderman. "We want to keep that raw, live feeling that we captured on stage."

According to Gilderman, recreating the acoustic ambience of the concert hall in the recording studio is a relatively simple matter — an opinion that was shared by most of the engineers we spoke with. "Take Cheap Trick, for example," he explains. "They have their monitors very loud when they're on stage. The microphones pick up everything that's going on around them. So when we overdubbed [lead singer] Robin Zander's vocal parts, I just gave him an SM-57 and let him run around the room with it and sing. I simply pumped all the instrumental tracks out through the speakers in the room while he was in there singing. With that method, I was able to match up perfectly with the live tracks.

Making the overdubs match the *picture*, however, is a much greater problem. Vocal and instrumental phrasing has to be perfect. "The hardest thing about doing *Rock and Roll Tonight* was the overdubs," says Pat Griffith, "because they had to match visually. The director loved close shots of the bridges and necks of guitars, which meant note-for-note matching in the studio. That took the longest for the bands to master. Bands that can barely play in the studio — we won't mention any names — all of a sudden had to deal with precision playing. They never play the same thing twice. Read music? Are you *kidding*?"

Another issue in matching audio to picture in concert shows is perspective mixing. "You can't mix a concert video like you would a record," says Griffith. "When you show the guitarist, the guitar track has to be way up; when the pianist comes on the screen, the piano track has to come up. You do not do a proportional mix in this case. If you were to listen to it on headphones, without looking at the picture, it wouldn't make much sense."

The problem is that perspective mixing might not be feasible every time for the simple reason that the cut picture isn't always available for the audio mixer's use. "If you have to mix blind while you're editing, then you can't really do a perspective mix," says Andy Zall. "So some people — depending on what kind of time crunch they're dealing with — can't do it. But it is the sort of thing astute producers have been asking for and accomplishing, wherever possible, for years."

"A lot of times, they're editing the show while you're mixing down," adds Brad Gilderman. "So even if you have a picture, you don't know if they're going to change their edits in the end. You sort of have to guess at what their shots are going to be. Obviously, you would want to push a guitar solo or something like that, because the chances are very good that they will be showing the guitarist during the solo."

Mixdown Formats

Final mixdown formats for concert videos differ, of course, according to the different media in which the show will be released. As a rule, though, several alternatives will be prepared in order to accommodate the possibility of future release in different media. At Complete Post, the final audio mix for concert shows is handled in one of the facility's audio sweetening rooms using a Quad Eight Coronado Console, Otari MTR-90 multitrack, Adams-Smith SMPTE synchronizer, and Sony BVU-800 for video playback.

"After the final picture is assembled,"

MOBILE AUDIO PRODUCTIONS

It would appear that through a production oversight, we omitted **Artisan Recorders** from our "Mobile Facilities Spotlight" published in the February issue of *R-e/p*. Artisan president Peter Yianilos tells us that the mobile's control room is equipped with dual MCI 24-track and dual MCI two-track machines, all with 14-inch capability and Dolby noise reduction. MCI and Midas consoles provide 40 fully equalized inputs. The wide selection of outboard gear includes units from UREI, Eventide, Ursa Major, Valley People, Orban, Inovonics, Roland, and MXR. Monitors are UREI 811As, with secondary speakers by customer request.

The complete remote facility consists of two vehicles: a 27-foot GMC Transmode, which houses the control room; and a separate support truck. "Among first-rate self-contained mobiles, none is more portable," Yianilos says, "enabling us to travel more economically as well as to situate the control room more effectively to suit a particular job. For instance, our clients have taken advantage of this by flying and shipping our entire facility overseas, and also enjoying very favorable travel costs over land. We have been permitted to park indoors at most major arenas, thereby reducing set-up and security hassles, and providing closer contract with the artist."

For further information, contact: Artisan Recorders, Inc. 1421A Southwest 12 Avenue Pompano Beach, FL 33060. (305) 786-0660.

explains Andy Zall, "it is brought down to the audio sweetening room — along with the rough audio track we used in preparing the final picture. The mixed-down multitrack tape [which has been prepared by the show's music producer] is also brought into the sweetening room and synced up with the edited picture. Normally, the video editor will give the audio mixer a list of the final audio edits, with the original timecode numbers. This way, he can match up, with the Adams-Smith, the exact playback numbers with the new master. In effect he's basically re-making the audio edits that were made in video editing [that is, to the rough audio tracks]. Only he's doing it with the original mixed-down version this time."

"Sometimes, they won't even have a fully mixed-down version," Zall continues. "They will mix it as they go along, making the final edits. Or, a lot of times they'll come in with a tape that has been submixed from 24 or 48 tracks down to four or five tracks. In the case of the Fleetwood Mac concert, they were down to six submixes — basically, each of the different voices, percussion, and the instruments. They could have set all six of those tracks at the same level, just run it, and it would have sounded fine. But, in this case, they wanted to play with the final levels a bit more.

"We have a lot of outboard effects in the room, which record producers and engineers will usually use to augment what they've already done with the mix. If the equipment wasn't there, they wouldn't ask for it. But since it *is* there, it gives the audio people one more chance to fix things up after they've been away from it for a while. That's something they really appreciate."

For shows like the Fleetwood Mac HBO concert, generally three mixes are prepared for layback to picture. According to Zall, "a two-channel stereo mix will be laid back to the one-inch picture master. Then we make a video copy and lay back a mono mix onto that. In addition, we made a four-track version which has two channel of stereo audio, a blank channel, and timecode from the edited master. That version will be used to make videodiscs, videocassettes, and things like that."

For *Rock and Roll Tonight*, which was simulcast on local FM radio stations, Pat Griffith mixed down the audio onto various tracks of a 16-track. "We would have a stereo mix transferred hot over onto two tracks of the 16-track," he explains. "On other tracks we'd have a Dolby-encoded stereo mix, a Dolby mono mix, and a non-Dolby mono mix. The audience tracks were transferred over separately from the mixed tracks — usually in delayed form. We put the audience on the original mixes, and then had a slap delay on two additional audience tracks that were roughly 0.9 seconds behind the original. The basic spatial delays equations involved gave the illusion of a big, roomy area. Finally, the timecode was transferred over onto a separate track.

"We therefore had separate mixes for the television listeners and for the simulcast listeners. The non-Dolby mono and stereo mixes were the ones we used. The final version of the tapes that went to the stations was C-Format — three audio tracks on one-inch tape: stereo right, left and mono."

Conforming Audio to Film Edits

While the procedures for shooting and editing concerts on videotape have been fairly well-established, film and tape hybrids remain a relatively unexplored area, due largely to problems in synchronizing the two media. For an HBO concert special with Linda Ronstadt, however, producer Robert Lombard devised his own system for shooting on film and editing on videotape, while avoiding any synchronization problems. The $750,000 piece was shot over several nights at the Arlington Theatre at Santa Barbara with a large orchestra conducted by Nelson Riddle. Ronstadt and her management insisted on using film, based on the results they had achieved earlier with Lombard on four lip-synced pieces for Cinemax's *Album Flash* program.

"In live concerts, you have a problem syncing up film to tape, or syncing the music up to the film," says Lombard. "Normally what people will do is shoot a live concert on film, cut a [rough mono mix on a] mag audio track [derived from] the 24-track master audio tape, and edit the whole thing on film. But on this project, there was no way we could take the time to cut on film — we *had* to cut on tape. Also, my director, David Lewis, is not a hands-on film editor and preferred to cut on tape as well. Immediately, Peter Asher and George Massenburg — Linda's record producer and audio engineer/sound designer — said 'It's never worked; no one has *ever* been able to sync up on tape after shooting on film'."

Lombard, however, had researched previous film-based concert projects — including the *No Nukes* film — made some tests of his own, and came up with his own variation of the methods used in those projects. Previous systems had called for each camera to provide a time reference for later synchronization by focusing in on either an appropriately-marked clapper board or a wall clock prior to shooting each song. Not wholly satisfied with these methods, Lombard, working with Randy Gladden of Rankin Corp., devised a system of nine video monitors bearing timecode numbers. These would be placed around the theatre for the cameras to focus on prior to each song. [A similar technique was also used by film director Hal Ashby during the shooting of The Rolling Stones movie *Let's Spend the Night Together*; see *R-e/p* December 1982 and February 1983 issues — Editor.]

. . . continued overleaf —

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"These video monitors," Lombard explains, "took two feeds from the Record Plant truck, which was recording the audio — a SMPTE feed and a 59.94 sync tone that in turn was fed into a 60 Hz converter unit. All cameras were referenced as usual to a 60 Hz sync source within each camera. The feeds from the truck went into a timecode reader and a character generator which, in turn, fed the SMPTE numbers into the video monitors. We fed them into the upper part and the lower part of the screen, so that each camera would hit the number twice, thus reducing the chance of mishaps."

Meanwhile, the Record Plant truck made a 24-track recording of the show, along with four-track and ¼-inch mono Nagra recordings bearing time code and a rough live mix of the 24-track recording. After the shoot, the four-track audio recording was laid down to one-inch C-Format video stock, with two tracks of audio on channels one and two, plus non-drop frame timecode on channel three. The film was then transferred onto the same one-inch stock, matching the timecode numbers that cameras had taken from the in-theatre video monitors with the timecode numbers from the audio tape. In this manner, synchronization was assured.

Off-lining was accomplished on ¾-inch videotape using Sony BVU-800s, which allow for stereo audio to be recorded in addition to SMPTE timecode. "You can't have a double feed of audio on conventional ¾-inch systems," Lombard explains, "and I wanted to have audio on *both* channels at all times. This meant I had to go to an address track system like the BVU-800, which means that you put your timecode right in the middle. Doing this also saved me anywhere from 10 to 20 hours in off-lining because the BVU-800s are faster than normal ¾-inch machines."

Final on-line assembly was accomplished on one-inch tape at Complete Post. While all this was taking place, George Massenburg and Peter Asher worked on overdubs with Ronstadt and her musicians. The project involved the usual compromises between visual and audio editing priorities.

"What we did was cut the show to what my director [David Lewis] felt would work visually," recalls Lombard, "but going along with the recommendations of the audio people — in certain cases, we had to use a particular night's performance. We found we could cheat a little bit, though, and cut in wide shots from other nights, and nothing would look like it was out of sync. We even got away with cutting in a few closer shots from other nights."

Lombard and David Lewis also shot several "visual overdubs" in a studio: closeups of Ronstadt and key musicians lip-syncing and playing various parts. Because they *were* closeups, no real background had to be matched, and the lighting also was relatively easy to recreate. Only here, instead of the singers and musicians having to match their performances to the picture — as in normal overdubs — they had to match their motion to the music. To facilitate this, Lombard relied on an elaborate system of "beep tone" cues he developed working on video clips with Ronstadt, Van Halen, and other artists.

In order to test the synchronization system through to the final phase of the project, Massenburg first made complete mixes for two of the songs, which were laid back to final picture edits Lombard and Lewis had made for those two songs. The results were satisfactory, and Massenburg mixed the remainder of the songs to picture after the picture editing was completed. The picture was then color corrected, and the complete, final audio, including audience and effects tracks, laid back to videotape.

In terms of actually integrating visual and musical material, then, concert videos offer many possibilities. Because the audio and video are prepared simultaneously, there is great potential for achieving a truly symbiotic relationship between the two — something that offers a distinct advantage over the promotional clip. The artistic uses of this potential have yet to be explored. The procedures for integrating concert audio and video could even lead the way to interweaving images and sounds created in a studio. As synchronization equipment reaches new levels of sophistication, and as film-video hybrid techniques become more prevalent, this exploration will no doubt begin to take place. ■■■

news

— *continued from page 22 . . .*

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The industry will feel the loss of Bill a great deal, not just for his technical expertise, but for his high standards of excellence, his integrity, and the outstanding example that he set for those who came in contact with him. He is said to have had one goal throughout his entire recording career: "To make the best-sounding recordings possible."

MORE NEWS *on page 161* —

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DAVID FOSTER

. . . "Mr. Fixit"

Interviewed by Robert Carr

Every industry or business has its share of "troubleshooters" — the specialists called in at the last minute to solve problems that appear to defy the strongest powers and well-laid plans of ordinary people. Ironically, the term "specialist" often is applied to that person, who possesses the broadest and deepest spectrum of understanding in regards to a certain field. And so is the case with award-winning musician, writer, arranger and producer David Foster. A native of Victoria, British Columbia, Foster studied the Toronto Royal Conservatory of Music program for eight years before moving at the ripe old age of 16 to England, where he played in back-up bands for Bo Diddley and Chuck Berry. Following a brief return to Canada, and a stint as a session musician with the Canadian Broadcasting Corporation, Foster headed for Los Angeles with his band, Skylark, and a top-ten single entitled "Wildflower." After that success, the world of records opened up to him. First, as co-musical director/keyboardist for the *Rocky Horror Picture Show* production and cast album. And then as session player/arranger for such talents as Barbra Streisand, Rod Stewart, Dolly Parton, George Harrison, and Ringo Starr.

Since the beginning of 1983, Foster has produced #1 hits in the pop, black, country, rock, and adult-contemporary categories. In fact, he's been so successful that his unofficial title in studio circles around Los Angeles is "Mr. Fixit." In association for the last six years with engineer Humberto (Hum) Gatica, who is presently earning co-production credits on many of the projects, Foster has accumulated an impressive track record.

R-e/p (Robert Carr): You've earned quite a reputation for yourself as a musician, arranger, songwriter and record producer. Do you relate to one role more than any of the others? How do you see yourself?

David Foster: I like to see myself as a musician, because that's how I got started. From as far back as I can remember, my initial goal was to get to LA, and become a session player. Luckily, when I got here, I met the right people, and started doing sessions in 1974 or 1975. [Drummer] Jim Keltner helped me a lot. And a contractor named Frank Decaro hired me to do a Mac Davis session, which was my first big-name session. From there I got on a roll. The songwriting aspect didn't come until much later, and the producer hat was even later than that. I still have many goals to reach to say that I've arrived from every angle.

R-e/p (Robert Carr): So the multifaceted situation you're in now was an evolution, rather than the result of some master plan?

David Foster: Being a keyboard player allows me to have the whole band at my fingertips, so to speak, and there was a natural evolution into arranging from playing, and from arranging into producing. I wasn't particulary a leader when I was a kid, but I wasn't a follower either. I was a "do-er." In LA, if you're a do-er you can get things done. So I evolved from player to arranger to producer, with songwriter thrown in as a necessity for the people I was working with who were stuck but wanted their songs to be finished.

R-e/p (Robert Carr): What do you imply by the term "do-er"?

David Foster: From the moment I set down in LA, I had blinders on. I tried to shield myself from the negative aspects of LA and the music business. I kind of charged full-speed ahead and worked really hard day and night. I call that being a do-er.

R-e/p: What do you perceive to be the

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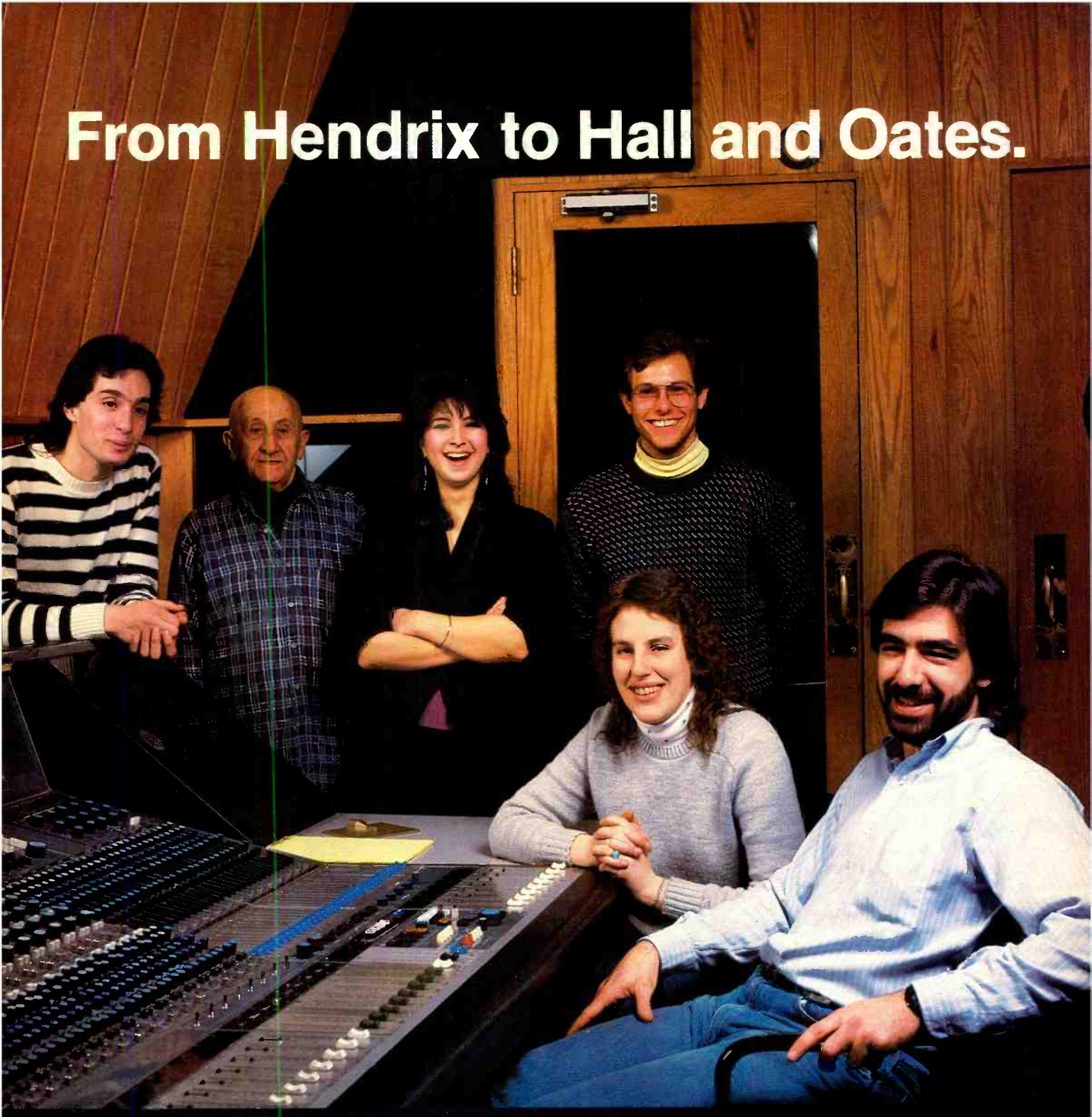

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DAVID FOSTER

negative aspects of LA?
DF: On a very basic level: Drugs. I know you don't want to do an interview about drugs, but I happen to be anti-drug, because I've seen so *many* of my friends go down from them; I've seen it ruin their careers. I don't go around bragging about the fact that I've *never* done drugs, but they're definitely a negative force in this town. I've been lucky to escape, and I think that is a short cut to success.

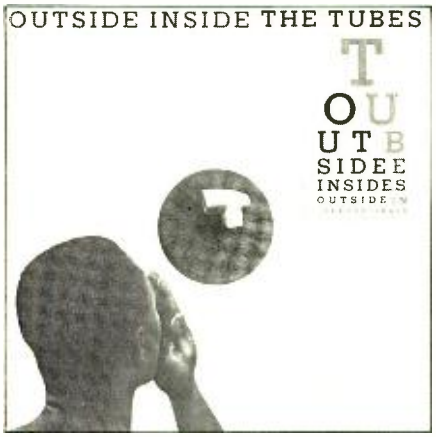
R-e/p: The attitude of most music people is that they like to think of other people in simple terms: He's an engineer; She's a singer; He's a songwriter. Sometimes wearing many hats can confuse the people that might be thinking of hiring you. How do people react to you?
DF: I do confuse people. It could be said about me that I am a Jack-of-all-Trades and Master-of-None. The very thing that gives me a broader appeal to artists is also the thing that hinders me, too. It's a Catch-22 situation. When an artist asks me to produce them, they're also getting a keyboard player for free, an arranger for free, and a co-songwriter for free . . . well, not for *free* [laughter].

The very hardest thing for me to do and I think that I've succeeded in doing it now — is to wear only the producer's hat in the studio. Before, I was a real "roll-up-your-sleeves, dig-in, play-every-note" type of producer. I'd tell the bass player every note to play; I'd work out the drum patterns — everything from top to bottom. I *don't* do that as much any more. It's much easier to produce when you can sit back and maintain an objective view.

I hired keyboard players to play for me, and I love it. It's wonderful for Humberto [Gatica] and me to sit in the control room as a team and have somebody else play the keyboard. It took many years to step down from being a musician *all* the time.

R-e/p: When you were on the player's side of the control-room glass, you must have run into some producers that you thought were misguiding the musicians, or just didn't understand the artist. Once you became the producer, were there certain things that you swore you would never do?
DF: That's such an important point. All of us — [Toto drummer] Jeff Porcaro, [Toto guitarist] Steve Lukather, [guitarist/producer] Jay Graydon, [session guitarist Lee] Ritenour, [session guitarist Larry] Carlton . . . all of us guys who have been doing sessions — tended to snub our noses at a lot of the producers. We felt that *we* knew how many notes there were in a G-sharp-9-flat-13 chord, and *they* didn't. Therefore, why were they the producers, and we were only the studio musicians? But now we're on the other side of the glass, we realize that they were *good* producers. They had the objective overall view, a love for music, and a good song sense. We didn't. All we have at that time was good musicianship.

When I first started producing, I'd cut a great track with the best players and then play it for all my musician friends. Everybody would go, "Man, that's unbelievable. That's going to such a hit." After that, I'd proceed to screw the record up totally, by putting all the wrong overdubs on, not caring about the vocal, not caring about the lyrics, not caring about the melody. All I cared about was the *music*.



"When you're in the top 20 you have some weight in this business; when you're out of the top 20 you have *no* weight at all."

I think Quincy [Jones] once told me: "Foz, an idiot can cut a good track. You just put out a chord chart and hire the best musicians in town, and you get a good track record. A track means nothing. It's the song, the melody, the performance, the mix . . . everything else."

So over the years we've been humbled, and we realize that being on the other side of the glass is *very* difficult. If that answers your question, I'm glad, because that's a really important question.

R-e/p: Since you're doing so many different kinds of projects — pop, black, country, rock, adult contemporary — how do you keep them all straight? How do you change or refocus your head to do them all justice?
DF: I began taking classical piano lessons at the age of five, and that gave me a firm foundation in classical music. I'm not saying that I expected to be a great classical keyboard player, because I realized at an early age that I was *not* going to be Van Cliburn. But I had 10 or 12 hard years of classical training, and that is the base for all types of music. Everything that we hear today, in one form or another, came from classical music, and that broad, broad base allowed me to shoot in any direction.

R-e/p: Obviously, there are stylistic differences that you have to get a handle on; there's almost a certain consciousness that you have to assume in order to handle a particular kind of music, such as R&B or rock, in the way it should be done?
DF: I don't know where the R&B thing came from. I grew up on Vancouver Island off the west coast of Canada, and there was *no* R&B or black music where I was growing up. I didn't even *hear* of Marvin Gaye until I was 20-years old. I grew up listening to the Beatles. Of course, they were pretty broad-based themselves. If you're going to get pop training, get it from the Beatles, because they had no barriers at all.

The other thing is: as you get better at your producing craft, you learn that if you just step back a little, and give the artist some breathing space, his natural ability will come out, and all you do is guide it. So why not make an R&B record one day, and a country record the next? If you like *all* forms of music, which I do, and you allow the artists to do *their* thing and just nudge them at the right moment, then there is no reason why you can't effectively produce all types of music.

I remember Paul Grein, in his *Bill board* column, "Chartbeat," wrote, "In the last year, Foster went to #1 on the country charts, the pop charts, the R&B charts, and on the easy-listening charts with four different artists and four different songs." I read that and thought, "Gee, that's pretty happening." I never even thought about it up until then. But there I was on all four charts within a year. I guess that could be taken two ways: "David Foster is pretty talented," or "David Foster doesn't know who he is; he'd better figure it out!"

R-e/p: So can we conclude from your varied track record that you need variety in your life?
DF: I guess I do. But there are other people, like Ted Templeman, who I just respect the heck out of. He stuck with the Doobies and Van Halen all the way, and they made hit after hit after hit together. That's one thing that is missing from my career. It's kind of happening with the Tubes and Chicago; we're going into our third album with each group. But as far as taking somebody from obscurity to stardom, I haven't done that yet, and I admire people like Ted who have done that more than once. Or take Mike Chapman, who has only done it that way. I don't think he's *ever* taken somebody who was successful and produced them. They have always been unknowns: Nick Gilder, Blondie, Exile, The Knack. He went to #1 with every one of them.

R-e/p: When you're dealing with just

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successful artists, aren't they locked into a particular style that made them successful in the first place?
DF: Yeah. But that's okay if you happen to *like* that style. I've *always* loved Chicago; from the moment we first got together we knew that I was the right person for them. They were just a little off the track.

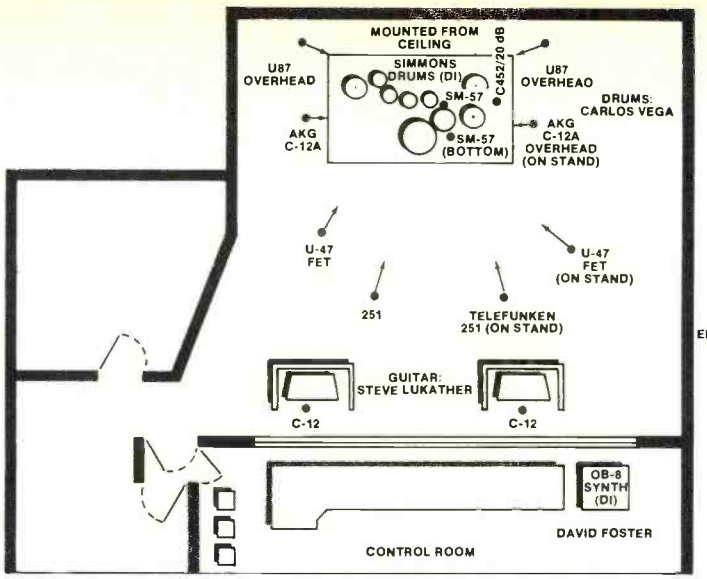
I looked at the Tubes and thought: "The only thing missing here is a hit. They're just around it, so let's zero in on a hit." One of the things I love about the Tubes' music is that everybody who hears it thinks I had nothing to do with it.

R-e/p: Everything you have produced that I've heard sounds like the group — not like the work of just one producer.
DF: So that means I have *no* style [laughter]. My style comes through my keyboard playing.

R-e/p: Chicago has done 17 albums now; that's a lot of music to have turned out. From the group's perspective, when it was planning album #16, wasn't there a need to bring in some fresh ideas, yet still remain faithful to its unique style?
DF: Always the problem with a band like that is how do you keep its sound, but bring it into the Eighties, and make it modern. The process *has* to start with the song. I felt that their last three or four albums have been song-light. Even though they are still capable of writing all those great songs they used to write, for one reason or another, they were off the track.

The essense of a producer, in my mind, is to be the Motivator . . . the Psychologist . . . who lifts the session up or brings it down when it needs it. Being a producer is a well-paying job that sometimes is very thankless. Other times it's *very* satisfying. The saying goes "If the record is a hit, the group gets the credit. If it's a stiff, the producer gets the blame." To some extent that is true, but I must say in the case of Chicago and the Tubes, they have gone overboard giving me recognition for helping them get on the charts.

R-e/p: What kind of preparations do you go through to work with an established artist? Do you study the past albums, or disregard the previous recordings and look at the band only for what they are doing right now?
DF: I was very familiar with all of Chicago's records — I could sing all the horn solos from the first, second and third albums! In the case of the Tubes, I knew "White Punks on Dope," "Don't Touch Me There," and that was it. To this day, they have made three albums with me, and six before that. I have not listened to one cut of their previous albums; I know *nothing* about their past, and I really don't want to. I don't know if that is good or bad, but it has worked. Before the first album I did with them, I went up to San Francisco and heard their material. They had 13 songs, and I loved every one of them. That was *all* I needed to know.



DRUMS AND ELECTRIC GUITAR SET UP FOR BASIC TRACKS DURING OLIVIA NEWTON-JOHN "TWIST OF FATE" SESSION AT SUNSET SOUND STUDIO #2, HOLLYWOOD
ENGINEER: HUMBERTO GATICA
PRODUCER: DAVID FOSTER

Partial Discography

All songs written and/or produced by David Foster:
"Wildflower" by Skylark
"Twist of Fate" by Olivia Newton-John
"The Only One" by Lionel Richie
"We've Got Tonight" by Kenny Rogers/Sheena Easton
"Heart to Heart" by Kenny Loggins
"Hard to Say I'm Sorry" by Chicago
"Love Me Tomorrow" by Chicago
"Mornin' " by Al Jarreau
"After the Love Has Gone" by Earth Wind & Fire
"Wait for Me" by Hall and Oats
"Talk to Ya Later" by the Tubes
"She's a Beauty" by the Tubes
"Hold Me 'til the Morning Comes" by Paul Anka
"Jo Jo" by Boz Scaggs
"Look What You've Done to Me" by Boz Scaggs
"It's the Falling in Love" by Michael Jackson
"And I'm Telling You" by Jennifer Holliday
"I Am Love" by Jennifer Holliday
"Got to be Real" by Cheryl Lynn
"I'm Alive" by Neil Diamond
"Friends in Love" by Dionne Warwick-Johnny Mathis
"How You Gonna See Me" by Alice Cooper

David Foster also picked up Tony and Grammy Awards for the *Dreamgirls* cast album, and Grammy Award for "After the Love is Gone." □□□

R-e/p: I would think that working with a self-contained group like the Tubes or Chicago provides you with a certain luxury — the band is all there; you have all the songs; the band can rehearse as much as they want before going in to record. When they are ready, you put them in the studio, and just run the tape. Obviously, there's more to it than that, but generally it's a relatively simple process. Yet, when you work with a solo artist, such as Kenny Rogers or Lionel Richie, you have to write the songs, find the players, write out charts, etc. Which type of session do you prefer?
DF: More and more I'm getting to the point where I prefer to work with a solo artist. I get frustrated working with a band, because I expect too much from them musically. Then, when they don't give me everything that I'm looking for, I want to bring in outside musicians, and the band members get mad at me, which leads to a big fight and nothing is accomplished.

I think at this point in my life, I shouldn't produce bands anymore, because it becomes a tearing-down process. Being so demanding, I take away the confidence that the band has. I said before that I'm working on the Tubes' next album, but actually I'm working on a solo album for lead singer Fee Waybill. I'm not producing the Tubes anymore for that very reason — because I'm so demanding.

The band members have a right to say, "We want to play on our own record. We don't want to bring in guys from Toto!" Yet in this case, a couple of the Toto guys were instrumental in terms of musicianship and co-writing, which helped us with the hits ["She's a Beauty," and "Talk to You Later"]. I don't know *what* the answer is with the Tubes. Hopefully it will lock in with the right producer and blow our minds with a couple of hits. That would be great.

. . . continued overleaf —

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For additional information circle #27

DAVID FOSTER

R-e/p: I see you as sort of a pinch hitter in the music business.

DF: Take a C chord and call me in the morning! I really feel like that sometimes. Kenny Loggins calls and says: "I have three hours this afternoon. Can I come over?" In three hours we'll start three songs. He takes the tapes away, pieces then all together, puts lyrics to them, mumbles and stumbles, and he's up all night working with them. Then he calls me a week later and says, "Well, come on over. I want you to hear the three songs you just wrote!" And they're just great.

But the thing is, I do have a lot of facility at my fingertips. That helps people like Kenny Loggins, who is incredibly talented in his own right, but sometimes they need that nudge. I wonder sometimes when maybe someone will come along and help *me*; I'm always helping other people. It would be interesting if someone else could give me a nudge.

Well, the answer to that is Humberto, who has been my engineer and beside me on every project for the last six years, is fast becoming a co-producer. We've been side by side for so long, we know each other like a book . . . similar, but not to compare to, Bruce Swedien and Quincy Jones. They've become one. We're becoming that way.

The best benefit for me is that I don't have to be in the studio all the time. Some albums are taking up to 1,000 hours to complete, and I used to have to be there 100% of the time. Fortunately, that time is being cut down, because of Hum [Gatica]. Now I can be there only 90% of the time, which is 100 hours less. Hum has recorded vocals for me without my being there. He's done guitar overdubs without me; we edit over the phone with me being in Canada, and we learn from each other. I feel fully confident in his ability and that is a *real* luxury.

R-e/p: You've provided a lot of hits for various artists, and you continue to get calls to produce and collaborate. How much of your popularity is the result of your talent as a songwriter do you think, and how much is the result of an industry tendency to ride a winner?

DF: I feel that when you're in the Top 20, you have some weight in this business; when you're out of the Top 20, you have *no* weight at all. I swear that I do feel differently when I'm not on the charts than when I am on the charts. I mean, it's impossible to be on the charts all the time, unless you're Michael Jackson [laughter].

Seriously, I used to be really concerned about what people and the industry thought of me. To quote Kenny Rogers, "If you build yourself up, the industry will tear you down. If you tear yourself down, the industry will build you up." That's true, but it's really weird when you think about it. If you downplay yourself, you're going to get praise, and if you go around hyping yourself, you get pulled down. It's also true that this business *loves* to see people fail. And I admit that I am not above that thought process. When my friends are on the chart, I'm happy for them, but I'm wishing that I was on the charts with them. It's just human nature.

As far as what the business thinks of me right now, I don't know. I know that I work hard, and I'm getting really good at being a producer. But basically I lead a very sheltered life. I have a family that I love; I go to work; I come home; I play hard. I'm an airplane pilot, and I'm in the process of buying a small jet that I'll fly myself. I have my boat, and I'm acquiring a lot of other side interests. I work hard and I don't really concern myself anymore with what people think of me. I get the feeling that I'm hot right now, but not I'm positive. And I honestly mean that. I just do my best. If I can keep working for another 10 years, that'll be great. Or . . . if I could be like one of the few exceptions like Quincy, Tom Dowd, and Arif Mardin, who are over 50 and still making great records. What a thrill that would be if my career could go like that. But who knows. Actually, you can do only three albums a year, and if you have a 10-year career, that's 30 albums. That's nothing.

R-e/p: Doesn't doing just one or two tunes on an album put you in an awkward position? You're going into an album project cold with the assumption that you're going to produce a strong single. How does that affect the continuity of the album?

DF: I don't like doing albums that have more than one producer on them. I think you are absolutely right; sometimes it blows the continuity. But sometimes it *does* work. On the newest Kenny Rogers album, I'm doing two cuts. We had the nice hit with Sheena Easton. I've done other projects where I've recorded just two cuts — with Boz [Scaggs], "Look What You've Done to Me" for *Urban Cowboy*.

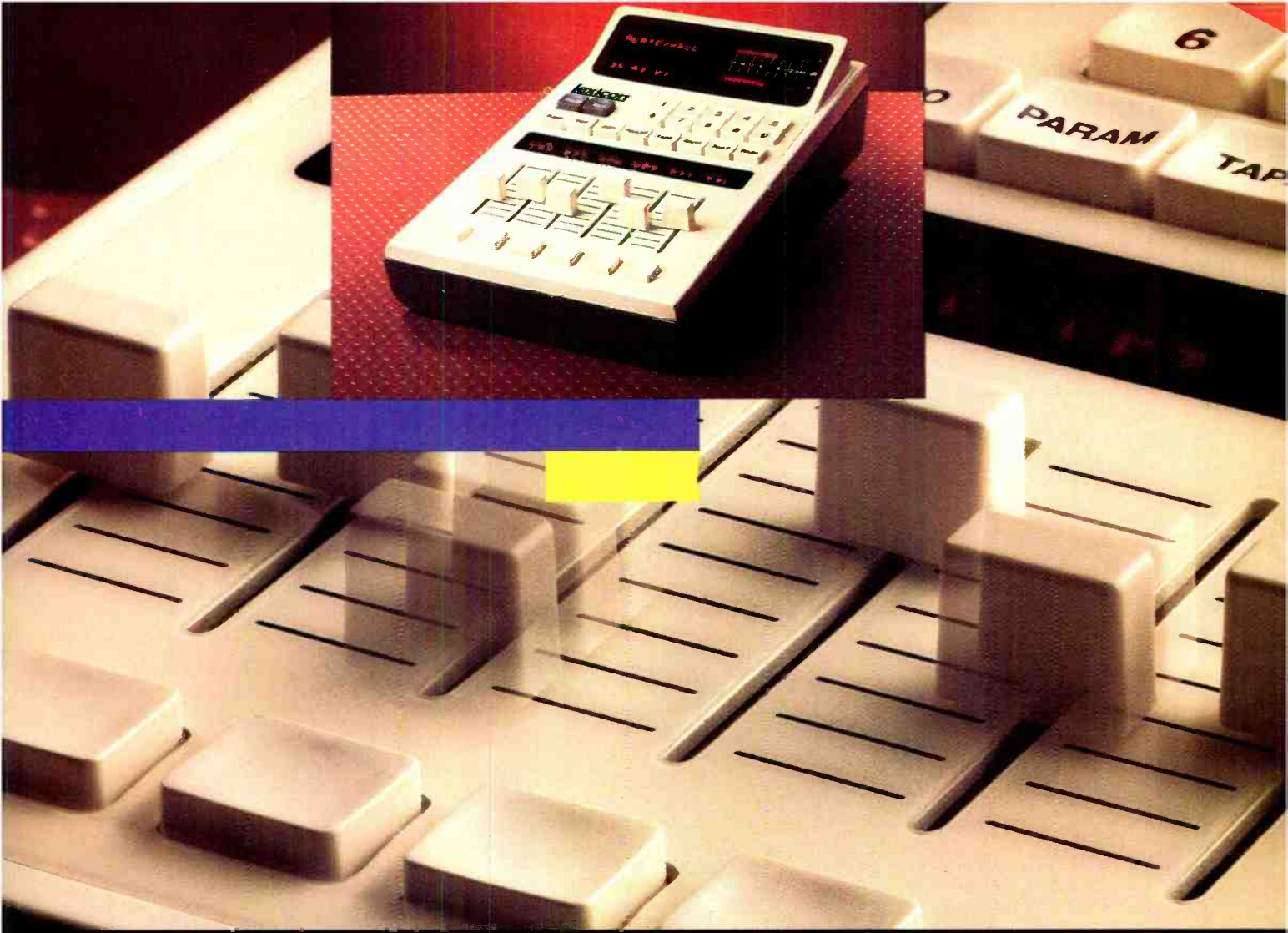
It's not so much that they come to me and say, "Oh, can you give us a hit?" because first of all that's not *always* true. And, second of all, the best arrangement is to have one producer for the whole album.

Usually, those individual songs happen because I'll be asked to do a whole record, and I just won't have the time. But it will be such an interesting project that one of us will recommend doing just a song or two instead. When you have only one song to concentrate on, sometimes that can be the single, because you've zeroed in so hard thinking about the radio. In the case of Lionel Richie's latest albums where I co-wrote and co-produced one song ["The Only One"], they're into their fourth single, and mine hasn't been chosen. So I'm definitely *not* always the guy who is brought in to come up with the hit!

R-e/p: If you're writing a song in that particular context, I would think you'd have to take into account the other songs on the album. Is there a skull session before the writing begins to discuss the album's concept?

DF: I don't know of any songwriter that doesn't say, "Let's write a song like so-and-so." Kenny Loggins is the first to admit that "Footloose" was patterned after some other tune. I would say that 95% of the time you pattern your song after something else you have in mind.

Sometimes when we get to the end of



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For additional information circle #28

DAVID FOSTER

an album project, we'll listen to the tracks, and there'll be one kind of tune missing. It happened with Chicago — "Hard to Say I'm Sorry" was the last song to be written, because I thought we have everything but a killer ballad.

Same with the Tubes' "She's a Beauty;" That was the very last song to be recorded. I said, "Guys, we need a song like 'Centerfold' by the J. Geils Band". Now I don't know if "She's a Beauty" sounds anything like "Centerfold" at all; it probably doesn't. But Steve Lukather, Fee and I sat in a room and I said, "Look, we've got to have a song that's a sing-along with rock-and-roll overtones." We ended up with "She's a Beauty."

R-e/p: Is there a list of criteria or types of songs that you want to include on each album?

DF: In my record making, I like to have one or two ballads, some medium tempo, and the up-tempo tunes, too. I was talking to Kenny Rogers yesterday about this. With some artists like Kenny, it's very difficult to do up-tempo songs, because they either sound contrived, or gimmicky, or out of context. That's why he sometimes ends up with a lot of middle-of-the-road, medium-tempo songs on his records. It would be *very* easy to overdo something that's up-tempo.

R-e/p: When I heard the Kenny Rogers-Sheena Easton duet, "We've Got Tonight," I got the impression that there may have been a problem settling on the right key for the song. Kenny sounded pretty relaxed and Sheena sounded like she was at the top of her range — kind of a mismatch of energies. What happened there?

DF: Very good observation, because that's *exactly* what happened. Sheena has such a high voice, and Kenny such a low voice. We cut that song in about three different keys. Finally, Kenny said, "I have to do the song in the key that I'm most comfortable in, and where I sound the best. If it's not right for Sheena, we'll have to find another singer." He didn't mean it in a hostile sense, because this was before she even came into the studio. But I had the feeling that the key was going to be too low if she sings it down, and too high if she sings it up.

Well, Sheena came in the next day full of enthusiasm, which was wonderful. I had never met her before, and was looking forward to it. While she was in the studio trying to sing to the track, Kenny and I looked at each other and went, "It's just too *high* for her. We'll have to get somebody else." All the while thinking that she would say, "Yeah. You're right. It's too high."

But she was so into doing it. "I'm going to practice this," she said, "and when I come in tomorrow, I'm going to blow your lights out, and make it happen." So the following day, we changed the melody around in a couple of places to make it work for her, and I think she sounds incredible. I think that explains what you heard in the record, but I don't think the average person hears that.

R-e/p: Returning to Michael Jackson, who we mentioned early, given the impressive number of hits that he's had from Off the Wall *and* Thriller, *do you think that word will come down from the record companies directing that they want more "Michael Jackson albums," with a half dozen or more hit singles?*

DF: No, I don't think that will happen. In another time, in another decade or another year, possibly all of those songs, given less of a chance, *wouldn't* have been hits. And I'm not saying that they aren't hits. The whole album from top to bottom is wonderful. But Quincy and Michael were on such a roll . . .

. . . continued on page 49 —

SETTING THE RIGHT MOOD: A TEAM EFFORT IN THE STUDIO

Creative rapport between producer David Foster and session engineer/co-producer Humberto Gatica

R-e/p's first question referred to the control room environment, and working together on a session: "How do the two of you function on a day-to-day basis?" we asked.

"Hum is becoming a co-producer," Foster offers. "It's not even that he learned it from me; we've learned from each other. His co-producing is better than my engineering, though, so he's coming out ahead [laughter]."

"It's kind of obvious that for the last 5½ to six years," adds Gatica, "I have been absorbing the way David produces. I admire what he does, so it is easy for me to put my feelings into the creative circumstance. Basically, we think pretty much the same."

"As I'm about to reach to turn up the vocals just a little bit more," Foster says, "he does it. Or about to turn up the Rhodes a little bit more in the 'phones, because the singer is singing a little out of tune, he does it . . . with *no* communications."

"It took a while," Gatica admits. "Two people can work together, even if they have two totally different opinions, and one of them, in a way, sort of learns to accept that. Maybe he thinks, 'I'm the engineer; this is my job. I'm going to make sure that the producer is pleased with what I do.' But then when he gets home he's saying, 'I wish I could have done it this way, but he [the producer] didn't let me.' Luckily, our situation is not like that."

Foster relates that when Gatica is working with other producers, he gives them 100%. "But when Hum's in the studio with me," Foster feels, "he's working *with* me — not *for* me."

And Gatica readily concedes that it took a while to get to that point. "David used to say things to me, and I would take them personally, instead of seeing the best of the comments at the time. Eventually, I learned. Now the [interactive] process is natural, and the comments never get in my way, because I look for what is the best for the project."

"That happens both ways," Foster continues. "When I'm overdubbing keyboards, Hum doesn't even hesitate to tell me what he thinks. He'll go, 'Don't play that! Hey, you don't want to play that'!"

Gatica says that his ability to offer creative input has evolved in the past year and a half. "But before that, I never wanted to insinuate anything to David Foster." Conversely, Foster says that he has now reached the point where he cannot handle keyboard overdubs

Producer David Foster (right) with engineer/co-producer Humberto Gatica at Lighthouse Studio, North Hollywood.

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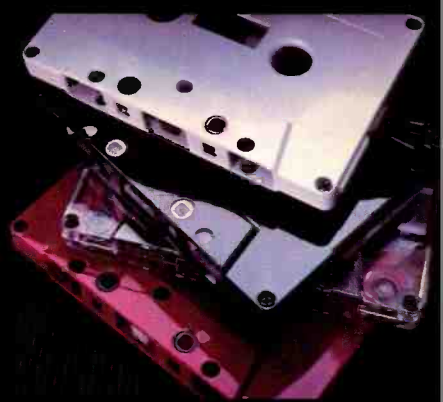

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SETTING THE RIGHT MOOD

— continued . . .

unless Hum is producing. " . . . Because he needs that objective ear," Gatica interjects.

"Like I said before," Foster offers, picking up the thread of his conversation, "I've worked so hard to wear just the producer hat, and not be a musician in the studio, that now when I'm a musician, I *need* a producer."

Assigning Production Roles in the Studio

Consider the following scenario, we offered: Let's say you have a contract to produce and record an artist. What kind of process do the pair of you go through to determine how and what you both will be doing on the session?

"Basically," Gatica replies,"the first time David mentions that we'll probably be working with such-and-such a person, I start thinking about it. I do my preliminary study so that when the day comes to .go in the studio . . . "

"For example," Foster interrupts, "when we found out we were going to do Fee Waybill's solo album, all I said one day was, 'It looks like we're going to go ahead and do Fee's solo album.' He said 'Great,' and nothing else was said about it. Then about three days later, Hum came to me with the Fixx album. He sat me down right here in this room, and played me each track of the album. Then he said, 'I've got it. I know what I want to do with this drummer. The key [to this project] is going to be the drummer.' Now I don't know if he spent 10 minutes thinking about it, or three whole days. But bingo, it happened."

Gatica offers that it is easier for him to relate to another project. "I don't want to copy the other record but that's the way this business is. Everybody — musicians, producers, engineers, etc. — borrows ideas from each other. At least, those people that want to learn do that. Those that are stubborn and want to do it one way don't want to listen to anybody else. We all go through that period of time where we don't want to accept the fact that other records sound *better* than ours."

"And engineers are the worst," Foster exclaims. "Here's what happens when you hire an outside engineer to do an overdub or something. The very *first* thing they say after they put up your tape is — and I swear this is true, they go: [Foster conjures up a disgusted expression] 'Wheww! *Who* engineered *this?*' It could be the greatest sounding track in the world, and they'll go, 'Let me just EQ this, and I'll make this sound . . . ' [laughter] "

Gatica admits that he cannot be paranoid about his engineering talents, and possible criticism of the sounds he creates. "That reaction comes when an engineer gets to the point [in his/her career] when you're confident you're a good engineer. The one thing that has affected my career is that I'm able to listen to other people. Inside of me I had to accept the fact that those other records sounded good, even *better* than mine. And once I did that, I was able to pick the best out of those records. My attitude became: 'What did they do in terms of sound, colors, balance or whatever to make this interesting' ? "

"Sound, a hit song and a good arrangement are all so closely connected," Foster says. When working as a session engineer, he could never understand why the music would sound terrible one day, and great the next day, despite the fact that he would be working on the same project, in the same studio with the same engineer. "Years later ," Foster recalls, "we realized when all the musicians are playing the right thing, they love what they are playing, the song is good, the melody is good, and the singer is excited about the song, the song automatically comes up a couple of notches."

"How many times," Foster continues, speaking to Humberto, "have you put the *exact* same mike on the *exact* same same guitar amp with the *exact* same guitar, and one day it sounds lousy and the next day it sounds great? There's no earthly reason for it . . . it's like voodoo or something."

Gatica confirms that, in certain respects, there is a close parallel between an engineer and a drummer that is playing a song he enjoys; he accents everything in a different way than if he was bored with the music — in other words, wanting to play it safe. "That conservative approach is not going to help the engineer on the other side of the board. The engineer wants to get inspired, too. I get excited about what I hear musically, and then I become more creative."

Drum Tracks — The Powerhouse of a Song

How do you determine what the drum sounds will be on the album, we asked? Gatica replies that, for the moment, sound is what making records is all about. "The sounds of drums have changed within the last two years. Everybody used to be going for the big, fat, deep snare sound, with the sound taking all the space in the mix, and completely covering the whole spectrum. Now, the trend is back toward the thin snare, and you create a big impact by using gimmicks and echo. It actually sounds really big, but it has more attack, and more definition, which provides more room for other instruments and voices to appear more up front."

"Ten years ago," Foster adds, "the snare drum was non-existent. There were rock-and-roll songs on the radio, where you couldn't even hear the backbeat. In fact, the backbeat was not even a part of the record."

"On some of the older records," Gatica continues, "the bass guitar was the major force. Not that that aspect has changed nowadays, but the backbeat has become more of an

DAVID FOSTER

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It's hard to say what is a real *bonafide* hit. By the time they got to the sixth or seventh single, they were coming on the charts at #20, jumping to #10, jumping to #6, and by the fourth week they were off the charts. What does that mean? To have a hit single in *Billboard,* you have to have two things: airplay, which he certainly was not lacking. But you also have to have *sales.* Everybody already owned the album. Why should they go out and buy the single? It became kind of a pointless exercise, and even Quincy admits that. Even though some of the tunes busted on, because of all the airplay, they couldn't stay on the charts without the singles sales.

The funny thing is that in your mind, while producing a record, you're trying

for quite a few singles. But you end up with only one, or two, or maybe three if you're *really* lucky. I don't think anybody is expecting me to make the next Michael Jackson record, although I'd like to sell that 35 million albums. Where are you spending the money, Quincy? [Laughter] It is a brilliant, brilliant album. The best of Michael Jackson, Quincy Jones, and the 62 musicians that participated — me being one of them!

R-e/p: Is the established artist, who has been cool for a while in terms of recent chart success, usually easier to work with and more open to new ideas than the acts who are currently hot?

DF: The word we use is "painless;" some acts are painless to work with. But in terms of recent successes on the charts, it's kind of unspoken. You don't talk about it. You just go at the current project with the attitude of "Here's the next record, and it's going to be the hit." If it isn't, you go at it again. "This is going to be the one."

R-e/p: If you have an artist that is difficult to work with, how do you approach that?

. . . continued overleaf —

SETTING THE RIGHT MOOD

— *continued . . .*

integral part of the overall track. I guess the old way of looking at it was: 'The beat is there so the backbeat has to be there, but let's not make a big thing out of it.' It's like the tennis player, who looks at the other guy's serve to see if he's a good player or not. The situation is exactly the same with music. If you don't have a good backbeat, right away you've approached the mix or the record in the wrong way."

So, in essence, you are both building the mix off the snare drum? "No," Gatica replies. "Actually, all the drums *are* there, but they're accented by the backbeat, because that's the first thing you hear. When a record doesn't sound very good, as an engineer I look at the technical aspects first. If you don't have a strong snare, the listener is usually not as interested, and it takes them a couple of more listens to see if there is anything good in that record — even if it's #1 on the radio. Some records become dated and boring, because the backbeat isn't that strong."

Studio Selection — Bigger may not be Better

Moving straight along to the recording and mixing environment, we asked: How big a part does the choice of the studio play in creating the sounds on a record? Foster is the first to answer: "We've learned that you *don't* have to have a $3 million, 'State-of-the-Art' studio to make a good record. I did a solo album right here in this room [the den of Foster's house] with a 24-track machine and a Yamaha PA board, and miked my piano out in the living room. Of course, we mixed it at a good studio with all the good equipment. But I defy anybody that hears it to say that it sounds like a home-spun recording."

"I still prefer working in a full-blown studio with all the best equipment," Gatica agrees, "but I'm to the point right now where if I had to make a session happen somewhere else, I would do it."

Foster says that they use a studio in the San Fernando Valley, north of Hollywood, called the Lighthouse. He admits that it is not completely state-of-the-art, but the room is very comfortable, the rate reasonable, and the owner, Eduardo, treats them like kings. "We never thought we could track there. We always used the Lighthouse as an overdub room, and tracked and mixed at the hot studios. Now we're tracking Fee's album there, and we love the sounds we're getting. Eduardo is also one of those very bright, technical people, so he's always helping us with the latest outboard gear."

But what is the bottom line that attracts the production team to a particular studio?

"Would the studios like it if we said that the bottom line is that it really doesn't matter?" Foster laughs. "It's not that it doesn't matter," Gatica adds; "it's just that we don't feel attached to the expensive places anymore. If you came to me, and told me you wanted a particular sound, I would *never* tell you that I could do it only at one place. 'I have no other choice; this is where I do my job best.' It's a valid attitude for mixing, because that's when you put everything together, and you must really relate to the tracks in terms of balance. But when it comes to recording, you can be more flexible if you know what you're doing."

For mixing, Foster concedes that he is partial to Kenny Roger's studio, Lion Share, which has Neve NECAM automation and Studer A800 lock-up capability. Next on his list is Sunset Sound. "We keep going back there. Actually, Lion Share, Sunset Sound and the Lighthouse are where we feel that we're part of the family."

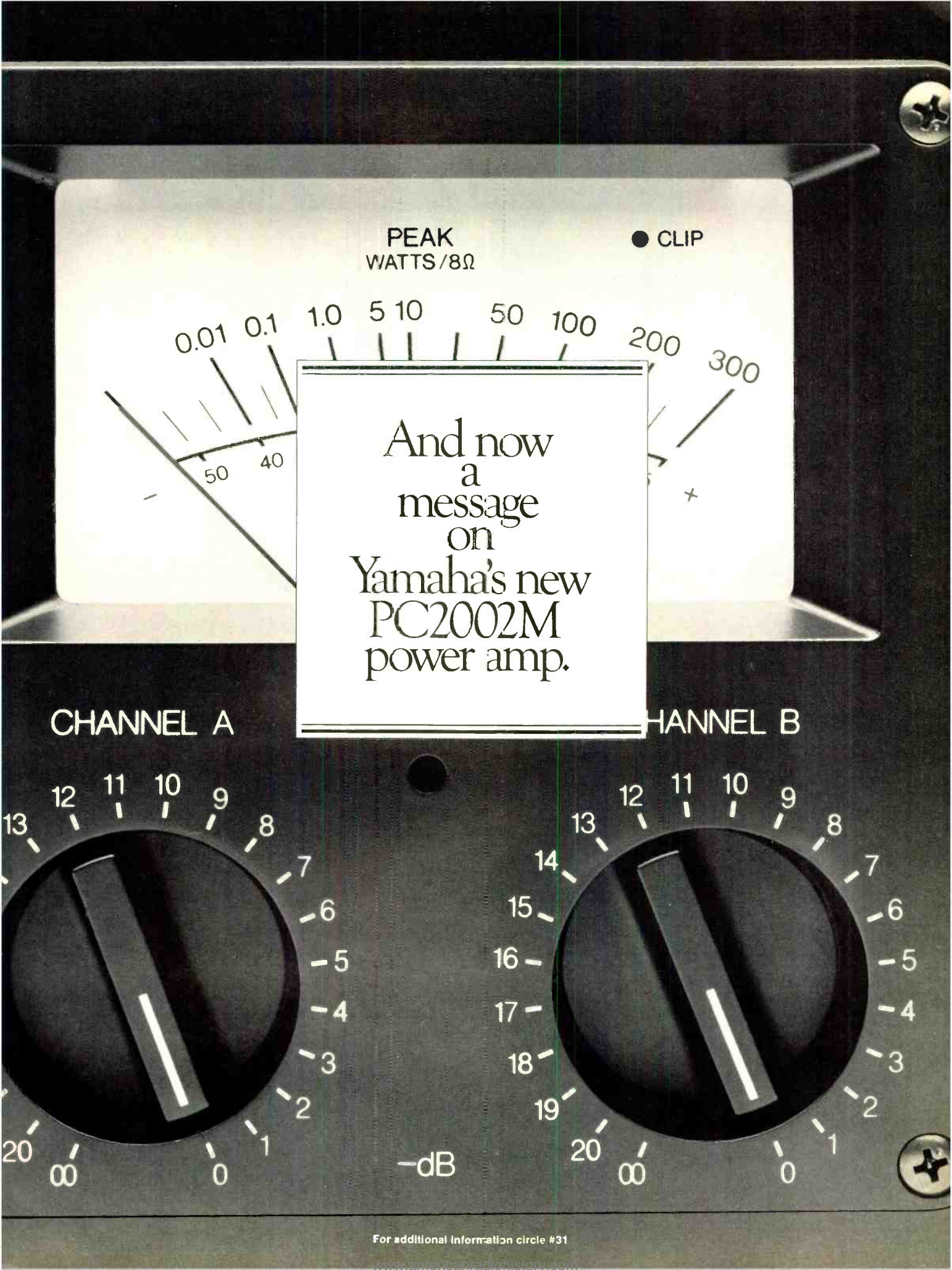
. . . *continued overleaf* —

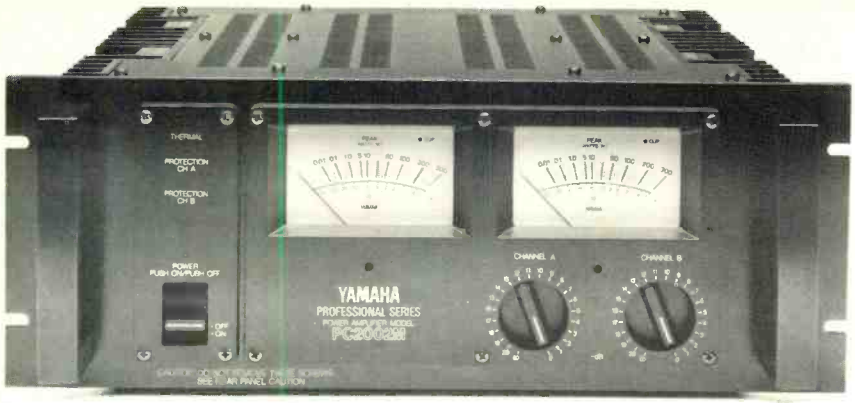
DAVID FOSTER

DF: I have a new way to approach that: I say, "No!" Right now I am at the point in my career where I really can comfortably say no. And if the act pushes the issue, I'll tell them why. Five years ago it was, "Yeah, walk on me. I'll do whatever you want," because I was new, and I wanted to get ahead. But it just isn't worth it anymore. I happen to be involved right now in two of the most painless projects that I've ever done: Kenny Rogers, and Fee [Waybill], the singer with the Tubes. They are both a delight from beginning to end. And my attitude is: I get up in the morning, and I really love going to work. It's just great.

The *Chicago 17* album, on the other hand, was *very* painful. It was seven guys who were fairly emphatic about how the record should be made, and rightfully so; they've had 20 hits. They deserve that right to have a *big* say in the way the record is made; *I* just happened to not agree with all of them. Being the producer, I had to put my foot forward, and demand that the record be made a certain way. Some of the guys are not happy about it. Who knows? If it sells three million copies, maybe they'll be happy. Or maybe they still won't be happy; I don't know. It was hard on everybody — on them and on me. Hopefully, the problems that we had will not show on the album. If those bad vibes seep on to the record, they will kill it faster than anything.

I don't mean to paint a completely dark picture about the experience. We've had a lot of success together. Peter Cetera is one of my three, all-time favorite singers in the world. The whole band is full of great musicians, and we all get





PC2002M

SPECIFICATIONS

			PC2002/PC2002M
POWER OUTPUT LEVEL	Continuous average sine wave power with less than 0.05% THD. 20 Hz to 20 kHz	Stereo, 8 ohms Stereo, 4 ohms Mono, 16 ohms Mono, 8 ohms	240W + 240W 350W + 350W 480W 700W
FREQUENCY RESPONSE	10 Hz to 50 kHz, 8 ohms, 1W		+0dB −0.5dB
TOTAL HARMONIC DISTORTION	Stereo 8 ohms 120W Mono 16 ohms 240W Mono 8 ohms 350W	1 k Hz 20 to 20 kHz 20 to 20 kHz	Less than 0.003% Less than 0.007% Less than 0.01%
INTERMODULATION DISTORTION	70 Hz and 7 kHz mixed 4:1	Stereo 8 ohms, 120W Mono 16 ohms, 240W	Less than 0.01% Less than 0.01%
INPUT SENSITIVITY	Input level which produces 100W output into 8 ohms.		0 dB (0.775 V rms)
INPUT IMPEDANCE	Balanced and unbalanced inputs, maximum attenuator setting.		25 k ohms
8 OHM DAMPING FACTOR		1 kHz 20 to 20 kHz	Greater than 350 Greater than 200
S/N RATIO	Input shorted at 12.47 kHz Input shorted at IHFA		110dB 115dB
SLEW RATE		Stereo 8 ohms Mono 16 ohms	60V/μsec 90 V/μsec
CHANNEL SEPARATION	8 ohms 120W 8 ohms 120W	1 kHz 20 to 20 kHz	95dB 80dB
DIMENSIONS (W×D×H)	18-7/8×16-1/4×7-1/4″ (480×413×183 mm)		
WEIGHT	PC2002 44 pounds (20 kg) PC2002M 45 pounds (20.5 kg)		

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YAMAHA

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DAVID FOSTER

"The essence of a producer is to be the Motivator . . . The Psychologist . . . who lifts the session up or brings it down when it needs it."

along really well outside the studio. But we had a lot of disagreements in the studio. I think that if they were sitting here, they would be nodding their heads, "Yes." But, like Quincy Jones says, amnesia does have a way of setting in. A couple of hits later it's, "Let's go back in the studio, and do it again!"

After eight months of recording and a month of mixing, the single came out and hit the charts at #49, their highest entry ever. If this album is really successful and we did work together again, I think it would be easier; they would maybe lighten up a little. Or maybe they'll say, "Now we know what to do, so see you later," and they may be very successful dealing that way. This is probably going to look terrible in print. It's just the truth, though.

But to get back to your question, I'll definitely be getting more selective. I'm trying not to work day and night, which is what I used to do when I was in my twenties.

R-e/p: Who in particular would you like to work with?

DF: If I said there is nobody out there that I'm *dying* to work with now, that would sound pretty egotistical, right? [Laughter] There are certain goals that I had, which have since fallen by the wayside. I always thought that I wanted to produce Barbra Streisand, and I never really pursued it. We are acquaintances; I have played on her album, but she has never approached me to produce her. And that has fallen by the wayside. Now I'm not dying to do it. If it came about, that would be great.

I always wanted to work with Earth, Wind & Fire, and doing the project with them was like a dream come true. And there is one other person: Burt Bacharach. From age 10, he was my hero songwriting-wise. And, to this day, I think that he is in that same category with Rogers and Hammerstein, and all them. He is one of the last living great songwriters. That makes him sound like he's old [laughter]. Burt, you're not old; you look great!! But he is of that calibre.

In fact, at the ASCAP Awards the other night they were honoring and awarding all these great songs, including a few of Burt's. Kenny Loggins, who was at the same table, turned to me and said, "They're doing this just to make us humble." We were sitting there thinking, "So what. We've written a few songs. Big deal. The whole world is not singing our songs. They're singing 'Raindrops Keep Falling On Your Head.' " So Burt Bacharach is my songwriting hero. I've written and played with him, and it was incredible.

R-e/p: You have a solo album coming out on Compact Disc and on an audiophile label, Mobile Fidelity. Are you planning on following that up with a solo career?

DF: No, it's just too much work. I don't know how some of these guys do it. They go on the road; they write their own albums; they do interviews every day of their life. I enjoy the other aspects of recording so much, that I don't think I can actively pursue a solo career. If it happens, great, but somehow I feel it's not going to just happen. There are too many other music-oriented projects that I want to do, and so many people that I want to write and work with. I'm really happy staying right here in LA. ■■■

SETTING THE RIGHT MOOD

— *continued . . .*

Digital Recording

Looking towards the future, had the team much experience with digital hardware? Gatica says that around the industry the buzz is that digital is the next step in recording, and he believes it will be. "But I don't think it is the solution now. You can still do good recordings using analog tape. We almost completed two songs on [a leading digital multitrack] but after two weeks of work on the machine, we heard the rough mixes of what we were doing, and didn't like them at all."

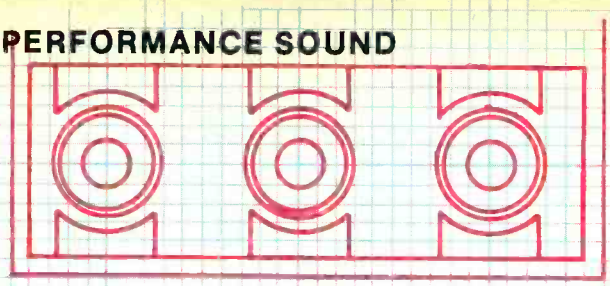
"We've since transferred the tracks to analog," Foster continues, "because we felt there were big holes in the digitally recorded sound. We should allow for the possibility that we could have had holes in our music. We're not used to working with digital recorders. Maybe we did something wrong, like equalization, or I played in the wrong register of the keyboard. There are so many unknowns."

Gatica realizes that they also have to acknowledge their established familiarity with a signal going through an analog board to analog tape, and that the analog medium introduces its own characteristics to the sound. "In order to get the sound we want out of the [analog] recording, we have to compensate for the changes through the use of equalization at the board, while the sounds are being recorded on tape. For example: when I'm recording a bass drum, I know exactly how much attack and coloration I'm going to lose or gain during playback. So I play tricks with the equalization, and that compensation process becomes natural to do."

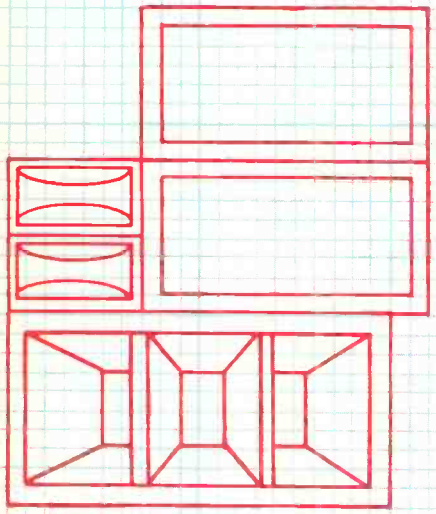
Foster admits that digital *doesn't* work that way. "What you put on tape is what comes back at you. But the fact is that the frequency response goes only so far, and then cuts off [at half the sample-rate frequency]. There is no sound beyond that point, no matter what."

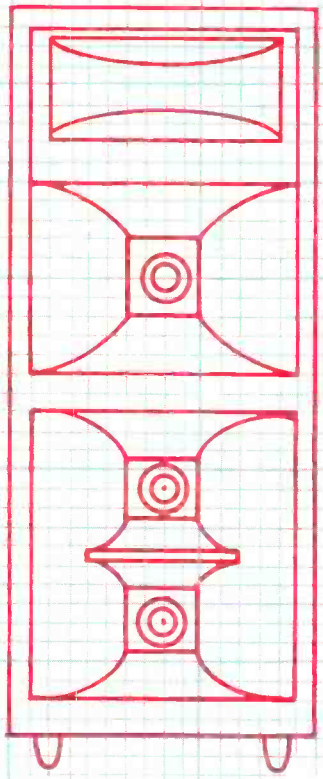
Gatica recalls that about two or three months ago he did a session with Kenny Loggins in which Loggins recorded his tunes on three 24-track analog recorders, and then pre-mixed and transferred everything to a Mitsubishi X-800 digital 32-track. "With almost 60 tracks to work with, it was a mad house, but the results were great. It seemed to work for me for a very interesting sound. So yes, we've tried digital, but at this point we're not ready to make the switch."

"We're not going to give up," Foster concludes. "We just don't know enough about it to make it work at this point. We're going to have to experiment more by ourselves for a while. Eventually, everything will be digital." □□□



RECENT DEVELOPMENTS IN CONCERT-SOUND SYSTEM DESIGN AND OPERATION





A Personal Appraisal
by David Scheirman

Over the years, *R-e/p* has presented numerous articles dealing with the design and operation of concert-sound systems. During that time, there have been many changes in both the requirements for live-performance sound, and the approach adopted by touring sound companies and equipment manufacturers in an attempt to advance the "State-of-the-Art" in arena and live-venue sound. This article concentrates on the innovative sound systems that have been described in recent issues by our live performance consulting editor, and is intended to provide an insight into the various changes in operating philosophies and design techniques of the profiled concert-sound companies.

A look at concert sound system technology in use today reveals that there is still as wide a variety of loudspeaker systems currently being utilized as there are touring sound companies to deploy them. Two decades ago, popular music groups used column-type loudspeaker systems when performing in concert, although horn-loaded sound system design technology was available from the cinema-sound industry[1].

The transition from columns to theatre-style, horn-loaded cabinets occurred during the Woodstock era. Prior to that landmark cultural event in 1969, most American musical groups owned small column-type sound systems. The fortunate few were able to invest in theatre-style horn systems, such as the Altec-Lansing A-7 Voice of the Theatre™ System.

In the pre-Woodstock era, musical events that involved a venue larger than a local high-school gymnasium were likely to rely on stacks of sound equipment rented from whatever regional musical instrument store was catering to the exploding "rock-band" market. Oftentimes these shrewd store owners covered the entire speaker stack with banners, thus making sure that the entire audience associated that store with their favorite rock group. As artist endorsement of sound equipment became popular, it was not uncommon to see guitar amplifier companies attempting to enter the PA marketplace.

As outdoor events such as Woodstock and Watkins Glen pointed out the need for better concert sound system, an integration took place between the equipment available to commercial sound contractors, movie houses, neighborhood rock bands, and professional audio engineers. During this time of hodgepodge sound systems, this writer vividly remembers seeing several different systems during the course of one summer (1968, I believe) and wondering... which system is best? The Jimi Hendrix concert featured stacks of Fender Showman cabinets piled high with University paging horns. The Who was carrying around Fender guitar cabinets for a PA, while Three Dog Night was singing through 24 Bruce speaker columns. The Beach Boys hoisted Sunn "Studio P.A." cabinets loaded with JBL speakers and horns in the air above the stage with tripods, while Mitch Ryder and the Detroit Wheels played to full houses with four Altec A-7s.

In the midst of this experimentation, a few individuals and partnerships in various regions of the country began to see that this new business of "Concert Sound" might turn into a worthy (if not profitable) avocation. The mid- to late-1960s saw the formation of companies organized specifically to service the

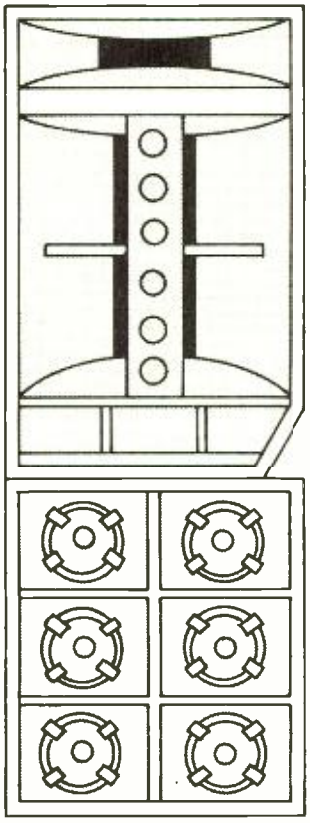
CONCERT-SOUND DEVELOPMENTS

rock groups that were touring across the country. Different sound companies tried out different types of loudspeaker systems then, and the same is true today. In general, however, most concert sound systems in use in 1984 are of the boxed type[2]; that is, due to logistical considerations, available truck space, and the search for more intelligible concert sound[3], the various loudspeaker components, after having sprawled across the sound wings in assorted piles of W-bins and long-throw horns, have evolved into modular packaged systems.

Some of those early sound companies that pioneered this business of concert sound are still with us today, and hold a dominant place in the touring market. Others have sold out, evolved into manufacturing or installation companies, or spawned newer, smaller, aggressive companies.

It is beyond the scope of this article to examine every type of concert sound loudspeaker systems in use today, nor do I wish to champion any particular type of system. As was true 20 years ago, different people like to hear different things. And different groups require different types of sound systems. Since recent articles in *R-e/p* have profiled nearly every basic type of concert loudspeaker deployment, it might be of interest to review the operational design philosophies of those sytems. Let's take a look, then, at an arbitrary cross-section of loudspeaker systems in use in the United States from 1982-84, along with updated system information and some designer comments.

Figure 1: Northwest Sound's 2-box system.



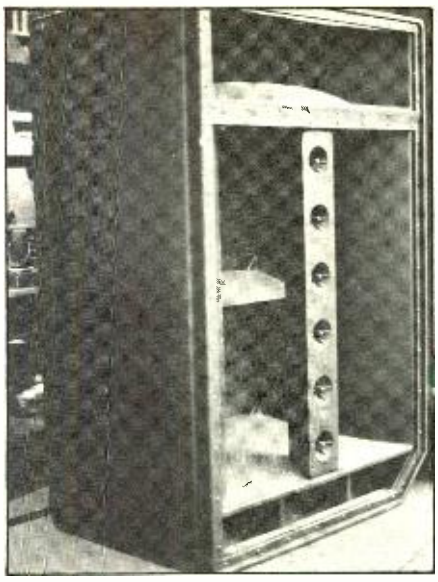


Figure 2: Northwest Sound's Model 590 loudspeaker cabinet.

GENERAL SOUND SYSTEM TYPES

The loudspeaker systems profiled may be grouped into four general system types as follows:

- ☐ Type A- *Stacked Component*
- ☐ Type B- *One-box Modular*
- ☐ Type C- *One-box Modular with Subwoofer*
- ☐ Type D- *Two-box Modular*

This general grouping delineates the packaging method used for various systems. Of course, a further breakdown of speaker system types could be made with respect to box design. Of particular importance in learning to differentiate between various concert loudspeaker systems is to recognize whether a system uses horn-loading or direct-radiating techniques for presenting the bass and mid-bass frequencies. A mathematical comparison of the two methods has yielded information that is applicable to concert sound system engineering[4].

Many contemporary entertainment-application loudspeaker systems easily exceed the required loudness criteria for mid-frequencies, but fail to provide sufficient output at low frequencies. A 4kHz tone is rated "very loud" by most persons at an SPL of 88 dB, whereas a 30 Hz tone requires a level of 118 dB to be judged as being equally loud by the same individual[5]. Horn-loading traditionally has been a method used by system designers to achieve a greater perceived loudness at low frequencies with fewer transducers and less amplifier power requirements. However, large bass horns can be quite bulky. Traditionally, folding the bass horn has been an acceptable compromise, while some of the more recent system designs strive to achieve an ideal balance between a direct radiator's lack of horn-throat distortion, and the economic realities of excessive capitalization for massive quantities of transducers. New cabinet designs, along with the newer generation of high-powered amplifiers offering power ratings in excess of 1,000 watts per channel, are making this an exciting time for concert sound system evolution.

The remainder of this article will take a closer look at the various concert-sound system this writer has profiled over the last two years.

MSI/NORTHWEST SOUND

Event Observed: Peace Sunday Festival at the Pasadena Rose Bowl (94,000 persons).

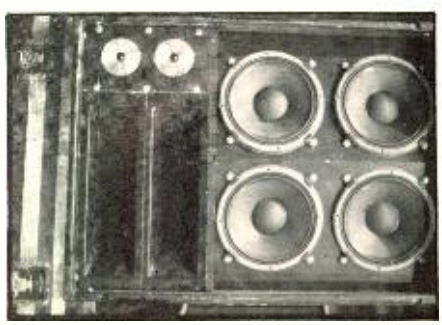
System: Type D combined with Type B (Tow-box modular and One-box modular) — see Figure 1.

In the early Seventies, Northwest Sound, based in Portland, Oregon, developed a unique loudspeaker cabinet that was particularly well-suited for touring concert sound systems at the time (Figure 2). Originally a two-way horn-loaded cabinet constructed of wood, the Northwest cabinet gradually evolved and incorporated such features as fiberglass horns, passive piezo-electric tweeters, and recessed wheels and handle for easy handling by one person.

"My first system was just like all the other systems in 1969," recalls Northwest's Bob Sterne. "You had truckloads of loose stuff...multicell horns, big W-bins. All of us in the business contributed little bits and pieces to what is now a 'State-of-the-Art' system. We realized that you had to gather up all the loose horns and speakers and consolidate things to be able to take the system around from hall to hall, and have the show up on time. Our low/high combination cabinet, with two 15s and a horn boxed together, inspired a lot of other system designs."

Northwest's Model 590 cabinet houses two 15-inch JBL 2220-B 16-ohm bass speakers, a JBL 2240 driver mounted on a Northwest Model 350 fiberglass radial horn with balsa-wood phasing inserts, and a vertical column of six Motorola tweeters. The box weighs 290 pounds, and measures approximately four by three by two feet. This cabinet, with its horn-loaded bass-reflex low end section, is intended for use as a full-range speaker system, or as the high-

Figure 3: MSI's High-Pack Cabinet.



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

BLUE NOTES
910 Business Loop
Columbia, MO
(314) 874-1944

Type of Venue: Club
Capacity: 500
Andy's Rating: **
Acoustics: Good.
House Soundman: Phil
Mixing Position: Center of house.
Console: Peavey
House PA: Peavey
Monitors: Mix from house (two mixes).
Stage: 22×16
Microphones: Shure
House Power: 100 amps.
Crew: Good.
Load In/Out: Easy through side doors right onto stage.
Overall View: Good rock & roll club; the Peavey PA was okay for this club.
Recommendations: Bring in own monitor and advance gig.

MISSISSIPPI NIGHTS
914 North First
St. Louis, MO
(314) 421-3853

Type of Venue: Club
Capacity: 500
Andy's Rating: **
Acoustics: Good.
Mixing Position: Center and in rear of room.
House PA: Yes. Small, not real loud/clear.
Monitors: Mix from house.
Stage: 22×16
Microphones: Shure, AKG
House Power: 100 amps.
Crew: Good and helpful.
Load In/Out: Through front door of club.
Recommendations: Bring in monitors & extra PA if band should be loud.

NICKS UPTOWN
3606 Greenville Ave.
Dallas TX
(214) 827-0561

Type of Venue: Club
Capacity: 550
Andy's Rating: ***
Acoustics: Good
House Soundman: Dale Trout; (214) 840-2976.
Mixing Position: Center of house 50 feet from stage.
Console: Yamaha PM-1000×16
House PA: Good JBL boxes; Sound Co. Dale Trout.
Monitors: Yes, good.
Stage: Small, 22×12
Microphones: Shure, EV
House Power: 100 amps.
Crew: Slow.
Load In/Out: Through doors, stage left.
Overall View: Ok gig.
Recommendations: Advance gig; get PA console up on a riser, get into club early.

NUMBERS
300 Westminster
Houston, TX
(713) 526-9119

Type of Venue: Club, large disco.
Capacity: 1200
Andy's Rating: ***
Acoustics: Good
House Soundman: Jim Caplin; LD Systems (713) 526-9119.
Mixing Position: Center of house.
Console: Whatever you need.
House PA: LD Systems; Art Donisa
Stage: 36×30
Microphones: Shure, AKG, Sennheiser.
House Power: 100 amps.
Crew: Okay.
Load In/Out: Good, loading dock right onto stage.
Overall View: Nice gig, lots of room.
Recommendations: Advance gig; get in early and make sure sound company brings what you need.

NIGHTLIFE
401 East Fourth Street
Austin, TX
(512) 472-4346

Type of Venue: Club
Capacity: 1000
Andy's Rating: ***
Acoustics: Good
House Soundman: Andy
Mixing Position: Rear of house way up high (not good).
Console: Biamp 24-by-4
House PA: Yes, good and loud.
Monitors: Good. 8 mixes with third-octaves.
Stage: 24×20
Microphones: Shure, EV
House Power: 100 amps.
Crew: Good.
Load In/Out: Loading dock, but all the way through the club down a ramp to stage.
Overall View: Good gig, but hard to mix from sound booth.
Recommendations: Just advance gig for yourself.

GRAHAM CENTRAL STATION
3301 Juan Tabo NE
Albuquerque, NM
(505) 292-5622

Type of Venue: Large Club
Capacity: 1200
Andy's Rating: ***
Acoustics: Very dead.
House Soundman: Ric Dustin.
Mixing Position: Center of house.
Console: Biamp
House PA: Show Tec Sound Co; Ric Dustin, (505) 299-5916.
Monitors: Yes, whatever you need.
Stage: 40×20
Microphones: Shure, AKG, Sennheiser.
House Power: 400 amps.
Crew: Good.
Load In/Out: Easy loading doors, up ramp to stage.
Overall View: Good gig, crew is helpful.
Recommendations: Advance gig to make sure you get what you need.

DEVIL HOUSE
430 N. Scottsdale Road
Tempe, AZ
(602) 894-0533

Type of Venue: Club.
Capacity: 800
Andy's Rating: ***
Acoustics: Good.
House Soundman: Neno
Mixing Position: Center of house.
Console: Yamaha 24-by-4.
House PA:: JBL 4-way system with BGW power amps.
Monitors: Yes, on stage mix 24×8 third-octaves.
Stage: 32×20
Microphones: Shure, Sennheiser, AKG.
House Power: 200 amps.
Crew: Good, big guys.
Load In/Out: Up a few stairs to club; not too bad; with good loaders.
Overall view: Good gig, PA was good, promoter good; good people to work with.
Recommendations: Advance gig w/Neno, house soundman; get PA stacks up in the air.

WOLFGANG'S
901 Columbus Ave.
San Francisco, CA
(415) 441-4334

Type of Venue: Club.
Capacity: 1000
Andy's Rating: ***
Acoustics: Good.
House Soundman: Kevin
Mixing Position: House left rear of room.
Console: Yamaha PM 1000-16
House PA: Meyer boxes with subs.
Monitors: Good onstage mix.
House Power: 200 amps
Crew: Good.
Load In/Out: Stairs, through double doors; slow.
Overall View: Good gig, PA was fine, good people.
Recommendations: Advance gig; get there early.

KEYSTONE
260 California Street
Palo Alto, CA
(415) 391-8284

Type of Venue: Club
Capacity: 750
Andy's Rating: ***
Acoustics: Good.
House Soundman: Jim
Mixing Position: Center of house; rear.
Console: Two Yamaha PM-700s
House PA: Yes; 4-way system.
Monitors: House monitors; mix from house.
Stage: 22×16
Microphones: Shure, EV, Sennheiser.
House Power: 200 amps.
Crew: Good.
Load In/Out: Back door and onto stage; not bad.
Overall View: Good club.
Recommendations: Advance gig; get monitors brought in if your act needs on-stage mix.

□□□

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

end complement to the direct-radiating Bass-Aug cabinet[6].

The Bass-Aug ("augmentation") box contains six front-mounted, 15-inch TAD speakers loaded in an infinite baffle. Although height and width measurements are the same as the other Northwest boxes, due to its increased internal volume, the Bass-Aug is three inches deeper. The Northwest cabinets are easily mounted together in vertical columns for flying systems, using nylon straps. Stage-level stacks are relatively stable, due to the Bass-Aug's depth.

Update: The combination in early 1982 of Northwest Sound with Maryland Sound Industries, of Baltimore, Maryland, has resulted in a further evolution for this type of system, and an integration of the two companies' design concepts. Recent Maryland Sound touring systems have been constructed along the Type D line (Two-box modular). MSI's three-way high pack (Figure 3) is coupled with the low pack (Figure 4) to create an integrated loudspeaker system, the boxes combining in building-block fashion to form hanging (Figure 5). The "low-pack" contains four 15-inch TAD loudspeakers in a tuned ported enclosure.

AUDIO ANALYSTS

Event Observed: The 1983 Styx tour, at San Diego's Fox Theatre (2,500 persons).

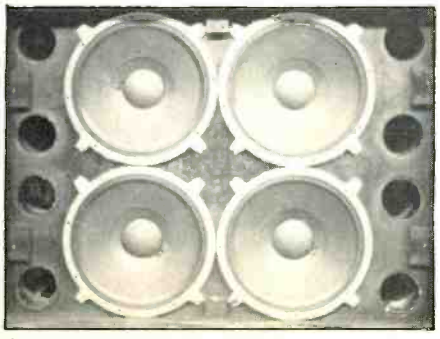
System: Type C (One-box Modular with Subwoofer — Figure 6).

Although this loudspeaker system is not typical of those normally used by Audio Analysts (based in Plattsburgh, New York, and Montreal, Quebec) as regards to cabinet manufacturers, it does fit the same type classification. Audio Analysts systems are most often seen with S-4 cabinets.

The Turbosound TMS-3 cabinets used on the Styx tour were purchased to satisfy personal preferences of the band's engineer. For the small-hall venues on this tour, approximately nine TMS-3s were stacked and flown per side, along with two TSW-124 subwoofers per side (Figure 7).

The TMS—3 measures 33 by 40 by 23 inches, and weighs 298 pounds when fully loaded with two LF 15-inch drivers, two MF 10-inch drivers, and a HF driver mounted on a 90- by 40-degree constant-directivity, foam-flare horn. The cabinet's low-frequency section is horn-loaded. The TSW-124 subwoofer cabinet is a foot deeper than the TMS-3, and not quite so tall. It contains a single 24-inch loudspeaker fitted with a four-inch voice coil, and has a rated power-handling capacity of 700 watts[7].

Figure 4: Maryland Sound Industries' Low-Pack cabinet.



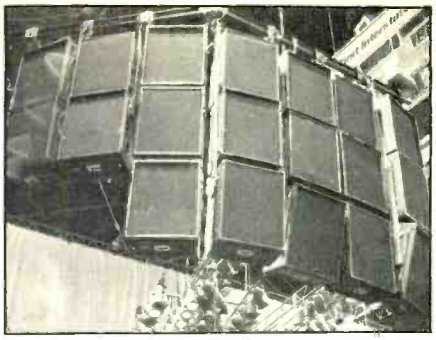


Figure 5: Maryland Sound Industries/ Northwest Sound's modular two-box system shown in flying-cabinet configuration.

For this particular Styx tour, Audio Analysts loaded the system with JBL 2225 15-inch loudspeakers, and JBL 2445 drivers on the high-frequency horns. "We had the cabinets shipped from England with only the 10-inch drivers pre-loaded," recalls Audio Analysts' Albert Lecesse. "We wanted to do some acoustical testing of our own to determine which transducers best suited our purposes."

Update: This type of loudspeaker system reportedly worked well for the Styx tour, which played to both small and larger arena-sized venues. The TSM-3 is one of the few high-level composite loudspeaker boxes offered by a manufacturer, though I have yet to personally encounter another major concert touring system using these Turbosound cabinets in the United States.

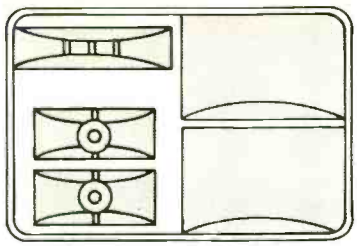


Figure 6: Turbosound TMS-3 cabinet.

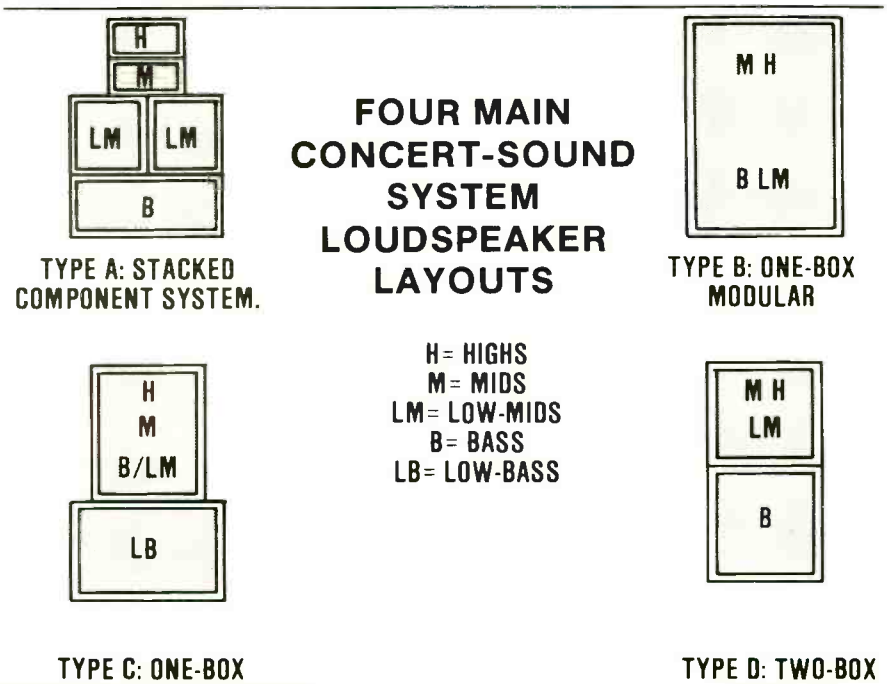
This particular system was certainly not the first to offer the subwoofer concept, but was indicative of a general trend to include low-end supplementary cabinets to composite-box systems that originally were designed to stand alone as a full-range cabinet. Audio Analysts continues to deploy its time-tested hanging arrays of S-4 cabinets, which are upgraded as advances in transducer technology take place.

ELECTROTEC PRODUCTIONS

Event Observed: The 1983 Roxy Music tour, at San Diego State University's Open-Air Ampitheatre (7,500 persons).

System: Type D (Two-box Modular — Figure 8).

In 1983, Electrotec Productions (formerly T.F.A. Electrosound) introduced a new loudspeaker array, known as the Lab-Q, which consists of a three-way active, two-cabinet system with a passively-crossed over high-frequency section. According to Pierre D'Astugues, Electrotec's senior VP, the Lab-Q system was designed with future trans-



ducer and electronics upgrading in mind as new technology became available.

The Lab-Q low-end cabinet is of a folded-horn design with center bracing and when first deployed, was equipped with two porthole vents (Figure 9). The LF cabinet contains a single 18-inch loudspeaker rated at 600 watts, and was developed for the company by JBL; frequencies up to 250 Hz are handled by this driver.

The second half of the loudspeaker system is a cabinet of identical dimensions designed to be stacked directly on top of the low-end box, special locking corner mechanisms being used to provide stability. The mid/high cabinet houses two, 12-inch JBL E120 loudspeakers that are loaded in a deep horn chamber (Figure 10). The E120s receive a signal bandwidth covering 250 Hz to 1.5 kHz. A JBL constant-directivity Bi-Radial horn with a JBL 2445 compression driver and two JBL ultra-high frequency units complete the package, which is rated at 600 watts per cabinet. The low-mid and mid-range horns together have an even horizontal dispersion angle of 60 degrees, according to Electrotec.

For the Roxy Music tour, 24 pairs of Lab-Q cabinets were supplied in the house system, giving the house engineer a total of 48 cabinets to cover venues ranging in size from 6,000 to 16,000 seats. For the show that I observed, the house speaker stacks were set up in vertical columns of three cabinets, seven columns per side. To adequately cover the 140-degree seating area, the stacks were given a slight hemispherical curve out to the sides and in towards the center of the house[8].

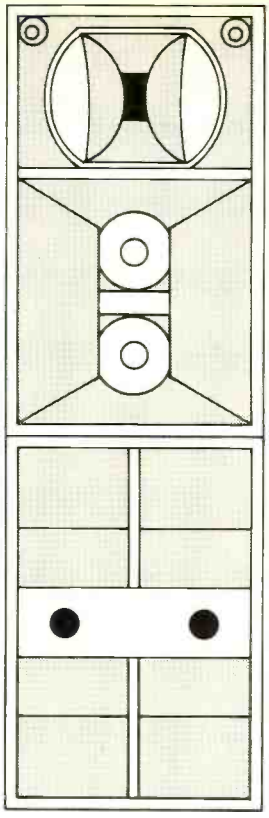


Figure 8: Electrotec's Lab-Q two-cabinet system.

Update: Over the last year, Lab-Q systems have worked on a regular basis on tours of all sizes. The system seems to go up and down fairly quickly. The company has designed the cabinetry to accommodate future design changes, while building the boxes of a very dense plywood for an anticipated long life expectancy. Electrotec is experimenting with both transducers and the low-frequency cabinet's turning to optimize system frequency response. Folded bass-horn technology apparently being used to an advantage to give the system a high perceived loudness factor with a minimum of loudspeaker cone area.

CLAIR BROS. AND SHOWCO

Event Observed: The 1983 US Festival in Devore, California (up to 300,000 persons).

System: Type B (Clair) and Type D (Showco) — Figure 11.

Clair Bros. Audio, of Manheim, Pennsylvania, and Showco, of Dallas, Texas, cooperated to provide sound services for both the 1982 and 1983 US Festivals, events that proved to be amongst the largest such festivals ever staged.

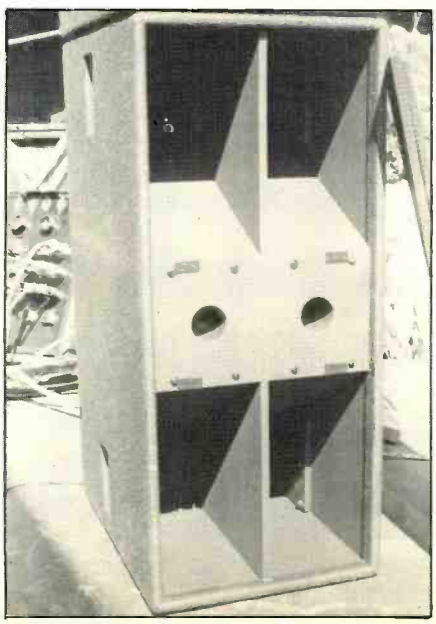
Since 1971, Clair has been building and using the S-4 composite loudspeaker cabinet, a design that was considered by many to be unique for its time, and still is the standard by which other composite cabinets are judged[9].

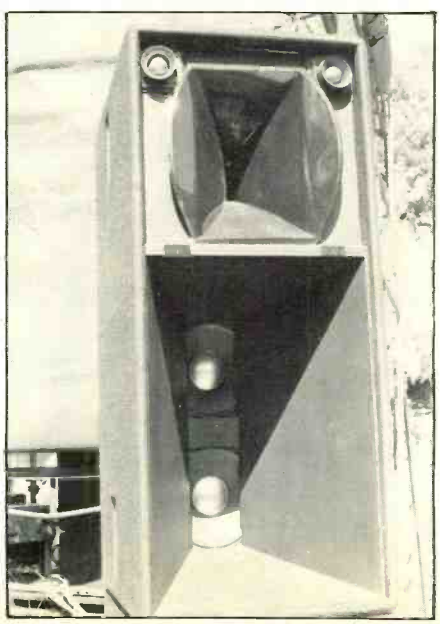
For the main speaker stacks at the 1983 US Festival, Clair provided a total of 180 S-4s, 90 of the 43- by 45- by 22-inch deep cabinets being stacked in 10 columns of nine boxes per side. Weighing in at approximately 425 pounds when fully loaded, this combination of S-4s gave the festival sound wings nearly 20 tons of loudspeaker cabinets to support on each side of the stage. Additionally, 40, 20- by 40-degree long-throw horns (manufactured for Clair by Community Light & Sound) were arrayed on the stack's upper level on each side.

"We were really pioneering a new concept when we first assembled the S-4s" noted Roy Clair. "We wanted something that would pack well in the trucks; that could be easily handled; that looked clean; and would give all of our systems a consistent sound. I think the all-in-one box [Figure 12] has proven itself to be the most efficient way to carry a concert system from city to city."

Figure 7: Audio Analyst's small-hall stack for 1983 Styx tour.

Figure 9 (left): Electrotec Lab-Q bass box. Figure 10 (right): Electrotec Lab-Q mid/high box.





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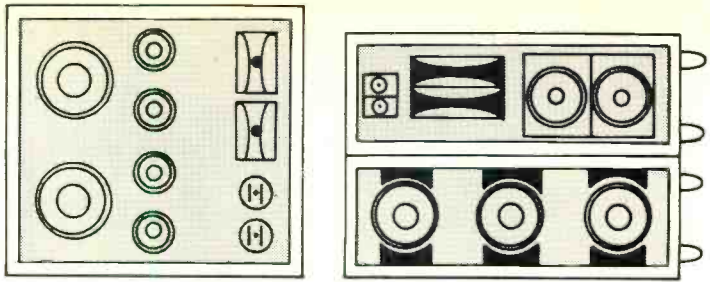


Figure 11: Clair S-4 composite cabinet (left), and Showco A.S. (Arena Sound) two-box modular cabinets (right).

Delay towers for the US Festival were provided by Showco. Each of four towers was stacked with 32 of Showco's A.S. (Arena Systems) cabinets: 16 bass boxes, and 16 mid-high boxes. The '83 US Festival marked Showco's first major deployment of the new system, which has evolved out of the famous Showco "Pyramid" system — a two-box, four-way setup that featured 300-pound cabinets each measuring 2½ by 5½ feet[10].

Extensive re-engineering and the inclusion of modern transducers provided the A.S. system with better frequency response characteristics, and shaved off some pounds. For ths US Festival, A.S. cabinets were loaded with three JBL 18-inch loudspeakers in the bass box; mid/high cabinets contained a pair of JBL E120 12-inch loudspeakers in short mid-bass horns, two JBL Bi-Radial 40-by-60 horns with 2441 drivers, and two Yamaha JA-4281 HF units.

Delay towers were located approximately 600 feet from the stage area, and the top of the A.S. speaker stacks was nearly 40 feet above the crowd (Figure 13).

Update: Clair Bros. has introduced two new loudspeaker systems for 1984. The S-4 Series II features the latest in transducers, including a new super-high frequency device which, according to Roy Clair, is "revolutionary," with a sensitivity rating greater than devices currently available to the public.

The new Clair C-4 cabinet has been deployed on the 1984 Yes tour. Dimensions are slightly larger than the previous S-4 box, and reflect the C-4's three, 18-inch low-frequency loudspeakers, and a horn-loaded mid-bass section with 12-inch speakers. For the Yes tour, the C-4 system is used in conjunction with sub-woofer cabinets. Clair systems are seen continually in use by major touring acounts in various configurations.

Showco's new A.S. systems is out with various major touring accounts, and represents a marked improvement over the older Pyramid system. Nearly 100 pounds have been trimmed from each pair of boxes. Riggers and stage hands with whom I have spoken this touring season report that the system takes less time to set up and is easier to handle.

DELICATE PRODUCTION COMPANY

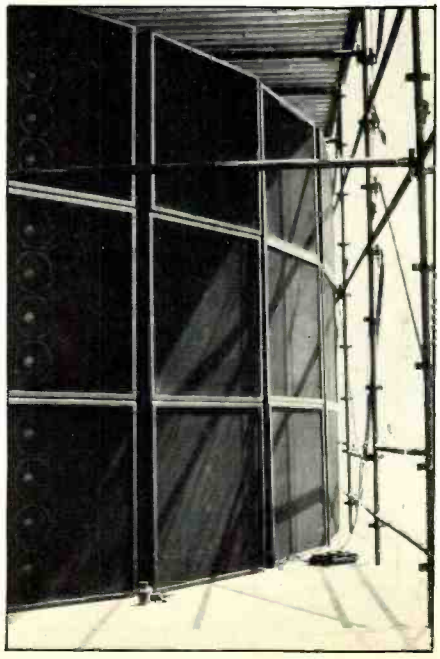
Event Observed: The 1983 Supertramp tour, at the St. Louis Checkerdome (20,000 persons).

System: Type A (Stacked Component) — Figure 14.

The Delicate Production Company's Supertramp rig began as a standard three-way, horn-loaded Martin system. Bass bins are the Model 215 cabinet, each containing two JBL K140 loudspeakers in independently-loaded horn chambers. This multicell exponential bass horn has a rated frequency response of 35 Hz to 1 kHz, and each cabinet is relatively small and lightweight when compared to some other bass cabinets loaded with the same number of drivers.

Low-mids in the system are handled by the Martin Model MH212 midrange horn, with two ATC 12-inch speakers that are compression-loaded into a 90-degree fiberglass horn. High midrange frequencies are reproduced with Emilar horns loaded with Renkus Heinz Model 3301 drivers. High frequencies are also handled by Emilar exponential horns, loaded with EA175 50-watt drivers.

Figure 12: Clair S-4 cabinets stacked for 1983 US Festival.



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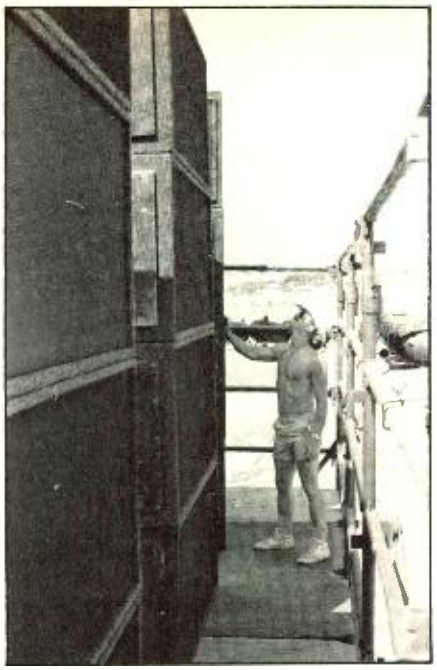


Figure 13: Showco A.S. system stacked atop the US Festival delay towers.

The Martin system is one of the few remaining stacked-component loudspeaker setups still in use by major touring acts. As Delicate engineer Mick Berg explained, "This system has always worked so well for [Supertramp]; I don't think anyone would even consider changing to a composite system. Our hardware and hanging baskets are set up so that the system goes up and down very quickly...we have no plans to change over anytime soon, but nothing is out of the question for the future[11]."

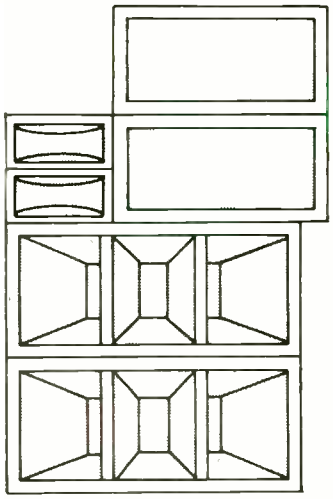


Figure 14: Delicate Production Company's stacked component system.

Update: It is sometimes claimed by advocates of composite-box systems that the stacked-component approach is outdated. Defenders of the separate component system concept point out that, while one-box systems may be less work to truck around, they are also *less* versatile when it comes to tailoring hanging arrays to specific venues. Componenet systems may point individual drivers where needed (Figure 15).

The Delicate Production Company's component systems are representative of a type of concert PA that evidently is still viable in the marketplace.

CARLO SOUND

Event Observed: The 1983 Oak Ridge Boys tour, at San Diego's Southwest College Stadium (8,000 persons).

System: Type B (One-box Modular with horn-loading) — Figure 16.

Carlo Sound's CS-3 cabinet was one of the first modular one-box speaker systems to use horn-loading. With a height of 6½ feet, the cabinet is not small. Weighing in at 422 pounds, fully loaded the CS-3 houses two JBL 2225 15-inch speakers, one 12-inch JBL 2202, and one JBL 2445 high-frequency compression driver on a Community Light & Sound Radical Radial™. First prototyped in 1978, the CS-3 was built for Carlo by Eastern Acoustics Works. The EAW B215 bass cabinet was the starting point for the CS-3's bass section[12].

For the Oak Ridge Boy's concert performance at Southwest College, Carlo engineers placed six CS-3s on each sound wing, along with a pair of smaller CS-2s on each side for added high-frequency coverage (Figure 17). The CS-3 boxes were seemingly built to last: steel plates for attaching hanging straps are flanged and mounted inside the

box's 18-ply birch walls. In addition, the cabinet's handles are not load-bearing points.

Update: After more than four years of touring with the Oak Ridge Boys, the original set of CS-3s still looks fresh; the cabinet apparently offers excellent structural integrity. A second system featuring CS-3s was assembled this year by Carlo.

"In addition to expanding our stock of CS-3s, we have developed a new, smaller cabinet known as the MH-118," reports Carlo's Rich Carpenter. "The new box has a smaller bass section, which houses an 18-inch speaker. It is 56 inches tall, 30 inches wide, and 24 inches deep, and works well for smaller touring systems as well as permanent installations."

dB SOUND, INC.

Event Observed: Stray Cats at the Del Mar Fairground's Bing Crosby Hall (4,000 persons).

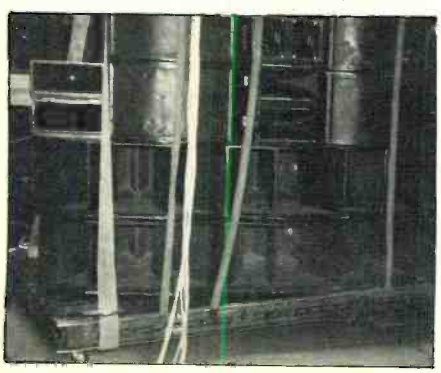
System: Type D (Two-box Modular) — Figure 18.

dB Sound, of Des Plaines, Illinois, has offered two-box four-way loudspeaker systems for many years. The loudspeaker system I observed on tour with the Stray Cats was dB's older-style system, which offered horn-loaded four-by-15 bass bins, which were slightly larger than the three-way mid/high boxes. The company's engineer's referred to a group of two bass cabinets and two mid/high cabinets as one "block," which took 3,600 watts of amplification.

Each mid/high cabinet housed two JBL E-130s in a horn-loaded, mid-bass section. McCauley Model 421 horns backed with JBL 2482 drivers and McCauley Model 428 radials with JBL 2425 high-frequency drivers completed the package.

"A two-box system such as this one is quite a bit more flexible in terms of setup than a one-box composite system," explains dB engineer Jeff Roeschlein. "And, it is a lot easier to travel with than a system which is made up of separate components. Of course, a system with separately-boxed components would be easier to stack, due to lighter weight, but I think it definitely goes a lot faster on setup and teardown this way. And your truck pack is much simpler[13]."

Figure 15: Delicate's flying component system, as used for 1983 Supertramp tour.



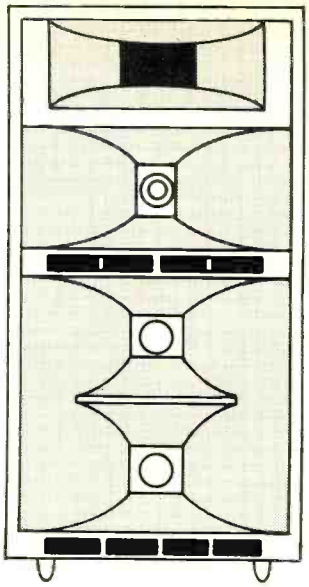


Figure 16: Carlo Sound CS-3 loudspeaker cabinet.

For the recent Stray Cats tour, four "blocks" were available per side; a total of eight bass and eight mid/high boxes per sound wing. In Bing Crosby Hall, dB engineers set up 13 cabinets per side (six bass and seven mid/high boxes). Cabinets on the outer end of each stack were placed in a vertical position. The system's engineers felt that the vertical dispersion pattern would help prevent excessive reflections from the narrow room's metal walls, which were in close proximity to the sound wings.

With 28 JBL 2482 midrange compression drivers packed into a relatively small venue which offered a highly reverberant acoustical field, I can remember finding this system to be extremely "loud." However, the excessive midrange energy seemed necessary to cut through the high ambient noise level of the crowd that was attracted to this particular event.

Update: dB's new HP-B/HP-3 loudspeaker system has reduced the size of the cabinets dimensionally by approximately 25%, and uses a 36-inch square-faced box that enables the construction of hanging arrays or stage stacks in building-block fashion. The HP-B (bass) and HP-3 (three-way mid/high) cabinets are identically-sized. Output of the new rig is reportedly double that of the company's older systems (Figure 19).

The new system uses four, 15-inch speakers per HP-B plus two, 12-inch speakers, two, two-inch compression drivers, and four JBL "bullet" tweeters per HP-3 cabinet. The first HP system was placed on tour with the Stray Cats in early 1984. dB Sound is building more of the HP systems for the 1984 touring season. In addition, a servo-driven subwoofer system is available for additional reinforcement in the 20 Hz to 100 Hz bandwidth.

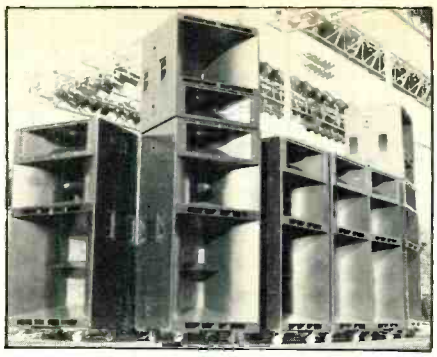


Figure 17: Carlo CS-3 cabinets deployed in stacked configuration for 1983 Oak Ridge Boys tour.

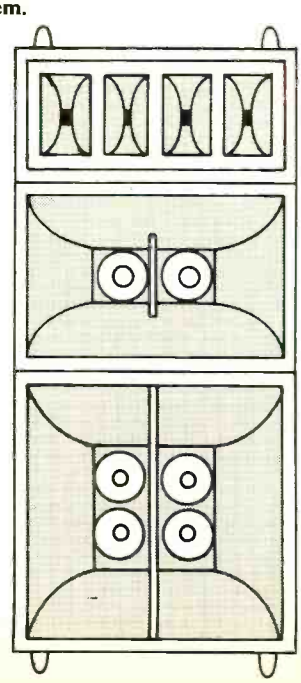
* * *

After reviewing eight different concert loudspeaker systems, it should be apparent that no single type of system works best in *every* application. The diversity available in system design is just as great as when I first started observing such systems in 1968.

Obviously, much has changed. Concert-sound systems are now truly able to be deemed "roadworthy." Materials technology and electro-acoustic research have helped bring about a variety of transducers much better suited to the concert sound business than those loudspeaker products that were available even a decade ago.

In reflecting on my impressions of the eight different concert speaker systems profiled in this series of articles, I was struck by the realization that the touring concert sound industry owes its very existence, in large part, to the failings of the commercial sound contracting community to respond to a changing

Figure 18: dB Sound's two-box modular system.



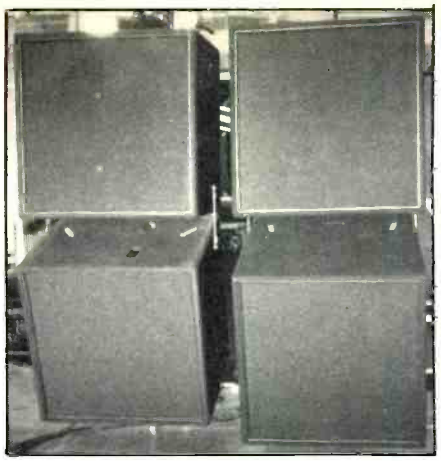


Figure 19: dB Sound's new HP modular speaker system.

marketplace. Degreed audio engineers oftentimes had no interest in the potential music-related applications of their own devices. State-licensed installation contracting teams merely hung in place what the blueprints said, rather than wondering if there might not be a better way to do it.

In upcoming issues of *R-e/p*, I will try to take a closer look at some of the concert sound companies that are starting to respond to changes within their own industry. Some of the new technologies being developed for touring concert systems are finding their way into other areas of pro sound: into installed house systems, and into products for general use.

One last thought: In hardly more than a decade, touring concert sound companies did an "end run" around the outdated arena and theatre sound systems by developing hanging sound systems, which brought high-quality, high-level sound to arenas and other venues not originally intended for musical performances. Will the farsighted venue owners then do an end run of their own around the large, concert sound companies by applying that very technology (along with some tremendous capitalization costs) to the upgrading of existing house systems, in order to be more compatible with contemporary musical events? I welcome dialog on this and other related subjects, in care of this magazine. ■■■

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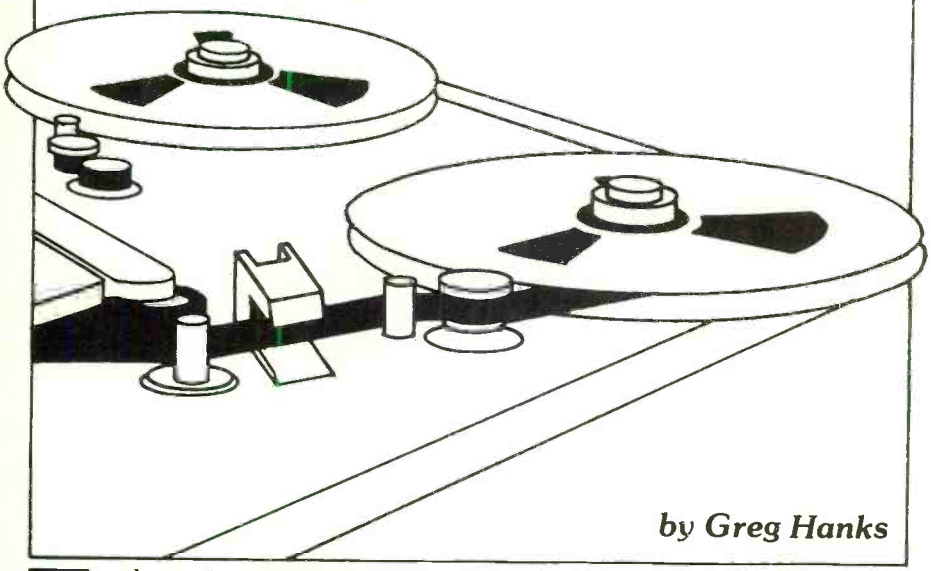
TAPE MACHINE CARE AND REPAIR

With a Special Emphasis on Transport Preventative Maintenance



by Greg Hanks

For those of you who were unable to attend the October 1983 AES Convention Workshop on tape machine care and repair, we are presenting here an overview on the philosophy of preventative maintenance, and the theory of tape machine operation. Although the basic concepts involved in putting sound on magnetic tape and getting it back off have been covered in extreme depth many times before in this magazine, we find that most of these treatises assume that the physical transport of tape is being handled, of course, in a theoretically proper manner! Most of us in the recording industry have been faced many times with multitrack machines that, because of mechanical malfeasance, are performing less than ideally. Some of the problems that I refer to are:

- Meter bounce at high frequencies;
- High-frequency instability that often times shows up as excessive noise;
- "Dirty" sound on sustained instruments;
- Incomplete erasure (no matter how many passes you make!);
- Speed instabilities, even though servo machines are in use;
- Etc. etc. etc.

The majority of tape machines will exhibit at least one or two of the above symptoms of mechanical distress at some point during their operating life. This article is intended to de-bunk a number of commonly held beliefs about tape-recorder maintenance, and provide the reader with a sensible way to approach optimization of the tape path. Let's start with some idea as to what the transport of a tape machine is *supposed* to do, and then we will discuss some methods that enable us to make the machine conform to our desires.

Optimal Transport Behavior

A tape transport is supposed to provide the following functions:

In the **Play** Mode —

1. Move the tape across the heads,
   A. At a constant velocity
      a) That conforms to industry standards, and
      b) With minimum short and/or long term speed deviations;
   B. At a uniform height
      a) That allows tape-to-head contact positioning which conforms to industry standards, and
      b) That imposes no undue pressures on the tape as to cause deformity of the medium, (for example, stretching, or scalloping of an edge or edges);
   C. At a uniform tension
      a) That allows for consistent tape-to-head contact, and
      b) That does not deform the tape, as above;
2. Transport the tape from the supply reel, and to the take-up reel,
   A. At a uniform height, and
   B. At a relatively consistent tension.

In the **Wind** Modes —

Provide a high speed transfer of the tape from one reel to another,

A. At a consistent tension
   a) As to provide a tight pack on the reel, and
   b) At a tension low enough to prevent damage occurring to the tape through deformation;
B. At a consistent height
   a) So as to be at a correct height on the reel for pay-out during the play mode, and
   b) So as not to damage the edges of the tape through scalloping or "long edge" deformation.

In the **Stop** Mode —

1. Provide sufficient tension to take up any slack in the tape path, and provide for tape-to-head contact;
2. No creep; and
3. Be sufficiently "soft" in tension to allow for manual manipulation of the reels for editing purposes.

Modern day recording adds some of the following demands to the above list:

1. Provide some means of measuring and timing the transported tape; and
2. Detect the end of tape and disengage the transport mechanism.

The above list of operating criteria sounds like it should be a fairly simple task to accomplish. Fat chance! There are a number of physical laws working against us, such as the little known law-of-gravity; we are trying to fight the gravitational force while transporting an elastic medium over fairly large, unsupported distances.

Theoretically, the ideal tape transport would operate with the tape path oriented vertically with respect to the earth. Not only would such an arrangement remove gravity from the guiding system, height of the tape path would also then be easily established by the use of edge guides. However, editing is not very easy when the tape traverses a path that is not perpendicular to the operator's line of sight.

In reality, what all current tape recorder manufacturers are producing is a machine that must provide all of the above mentioned functions, with the least compromise and for the minimum capitol expense. There are a number of difficulties in making tape recorders that satisfy all of the above criteria, not the least of which is economy of scale. When you consider that there are less than 15,000 24-track machines around the world, and you compare these numbers to say the manufacture of automobiles — roughly 15,000 a week per manufacturer — then you realize that the economy of scale simply is not there! This fact makes tape recorders, of the two-inch variety at least, *very* expensive. (The preceding discourse was for the benefit of those makers that we all denigrate for not doing their job as well as we think they should!)

On any orthagonally designed tape transport — in other words, one in which the tape path is at right angles to the deck plate — the primary tape-guiding element is the supply-reel height[1]. The objective of the transport's

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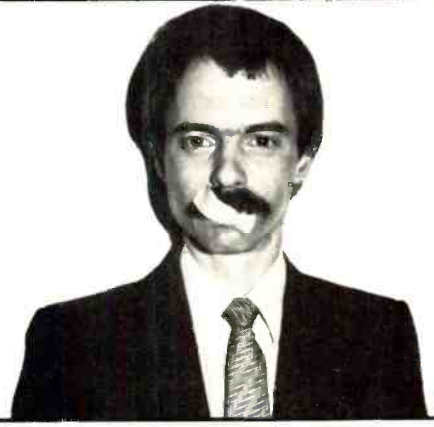

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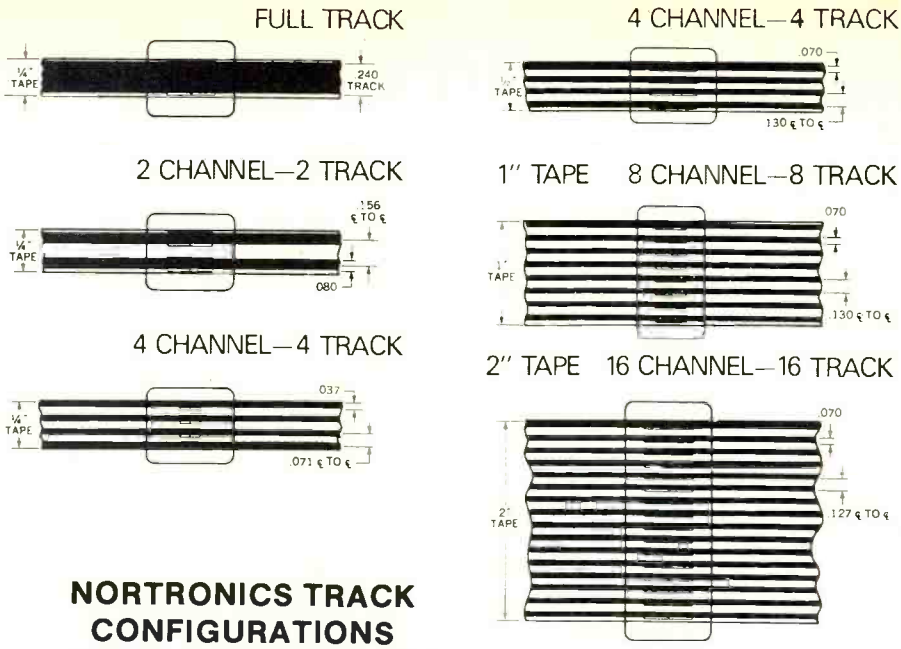
guiding elements is to move the tape past the heads, with a tension that is uniform across the surface of the tape. Without the supply reel being at the reference height, either the bottom or the top of the tape will have a longer distance to traverse and, therefore, have a greater tension than the opposite edge.

All transport systems have a designated tape path height above a given reference surface: this height must be the same throughout the tape path (unless, that is, you consider something like the videotape machines, where the tape must travel up or down a given path. In these circumstances, however, the supply and take up reels are angled so that the bottom of the tape, as it moves through the tape path, is a straight line). There have been some instrumentation machines constructed with the reels situated directly above one another, and the elasticity of the tape was utilized to accommodate a tape path that was not a straight and linear path. However, the short wavelength performance of these machines depends upon the consistency of the given elasticity constant of the tape; as a result, they perform poorly with high-frequency analog signals, using different varieties of tape.

All currently manufactured tape recorders intended for studio use are designed to move tape in one plane of travel only. It is a known phenomenon that when a differential tension exists across a moving elastic surface, there is a shift in the plane of travel, so as to equalize the tension across the elastic surface — an effect that accounts for the "hill and dale" action of tape in a transport. The tension can be made dissimilar between the top and bottom of the tape by any number of different aspects of the transport assembly. The most common variant is the supply-reel height. Any non-parallel guiding surface will also cause a top to bottom tension differential in the tape path, which implies that not only must all the motor spindles (capstan, supply, and take-up) be perpendicular to the line of tape travel, but also that all perpendicular guides (contact surfaces) be parallel to one another. (You can imagine what kind of havoc occurs when the supply reel motor shaft has taken a shot, and has 20 to 50 thousandths of an inch run-out at the top!) The edge guides must also be the same height above the reference surface, so as to form a straight line that is positively parallel to the bottom edge of the tape.

There are, however, some misnomers in the above statements. In order to transport the tape off of the supply reel to the first incoming guide, and fulfill the requirements outlined in our introduction, the supply reel flange must be a minute distance from the bottom of the tape. Gravity pulls the tape to the bottom flange of the supply reel on larger tape sizes. The flange must, of necessity, be below the reference tape height so that rubbing does not occur as the tape leaves the rotating flange. The distance that the reel is below the reference surface must be minimized.



NORTRONICS TRACK CONFIGURATIONS
(Values slightly different from Ampex chart below.)

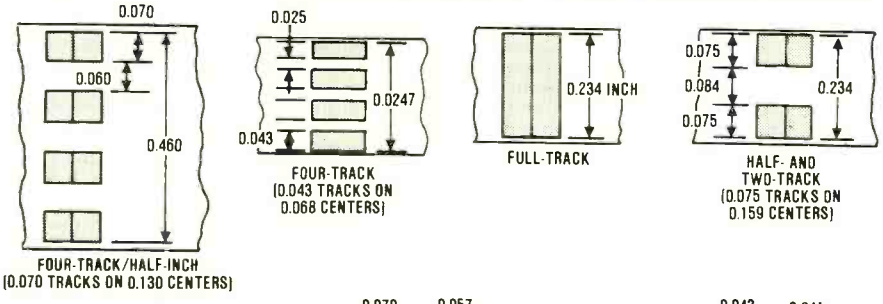
Tape Path Guidance

There are a number of different types of guides. The most common types are:

- Supply reel
- Fixed edge guides
- Fixed surface guides
- Fixed guides posts incorporating edge guides as a top and bottom flange
- Rotating surface guides
- Roller guides (with edge guiding)
- Roller edge guides (rare)
- Capstan(s)
- Pinch Roller(s)

The way that these various guides function is both similar and obvious upon inspection. The least obvious guide is the rolling surface guide. When the tape contacts the roller, it travels in such a way as to minimize any drag coefficients that may exist across the surface of the tape. This means that the

AMPEX TRACK FORMATS (Taken from Specification Sheets)



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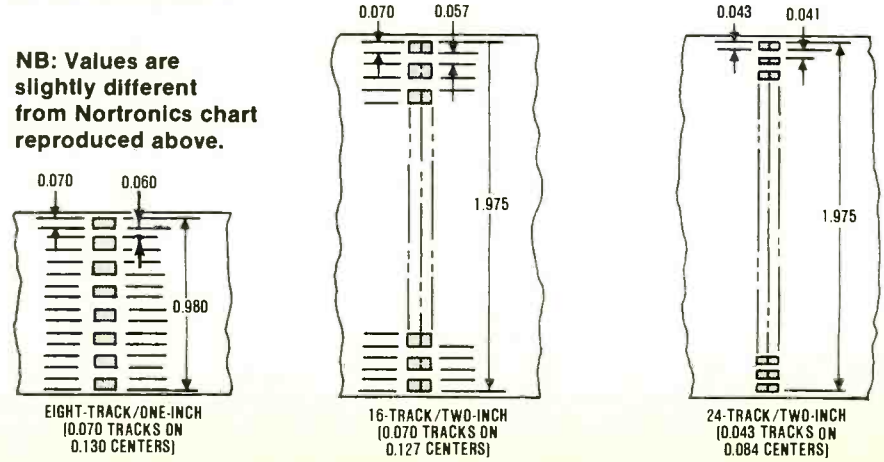

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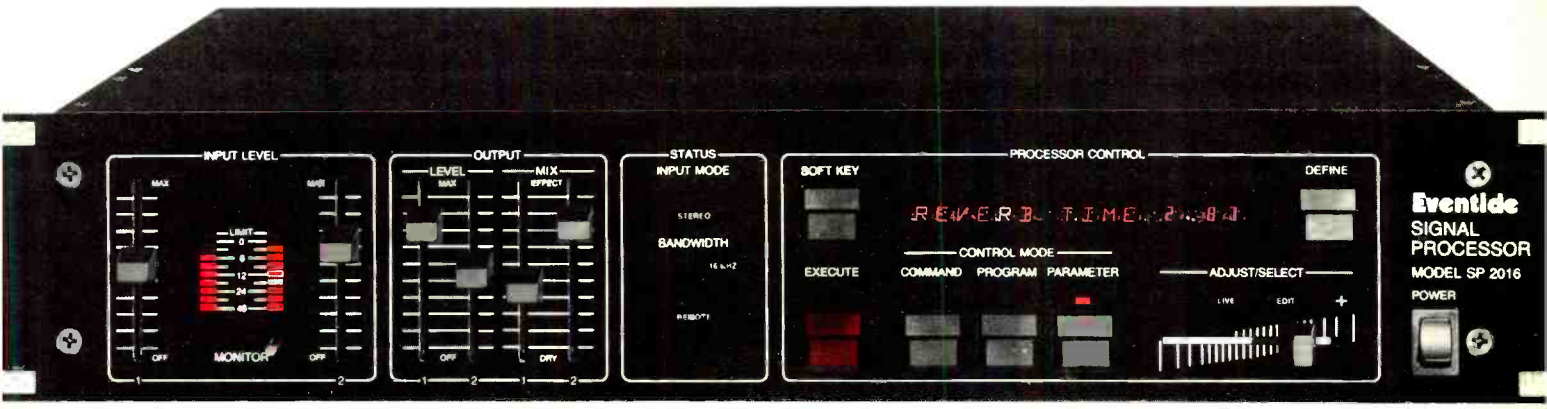

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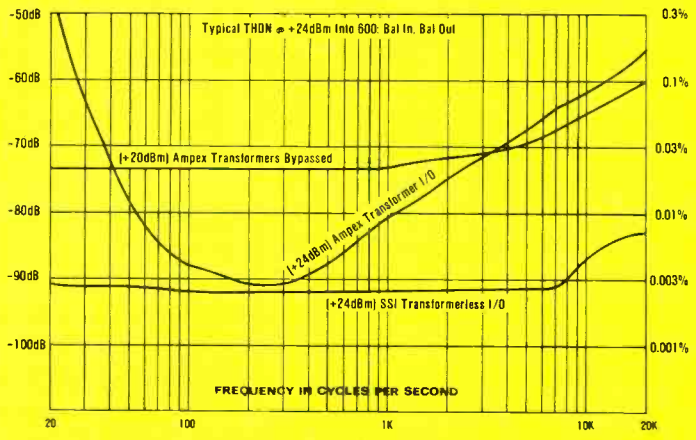

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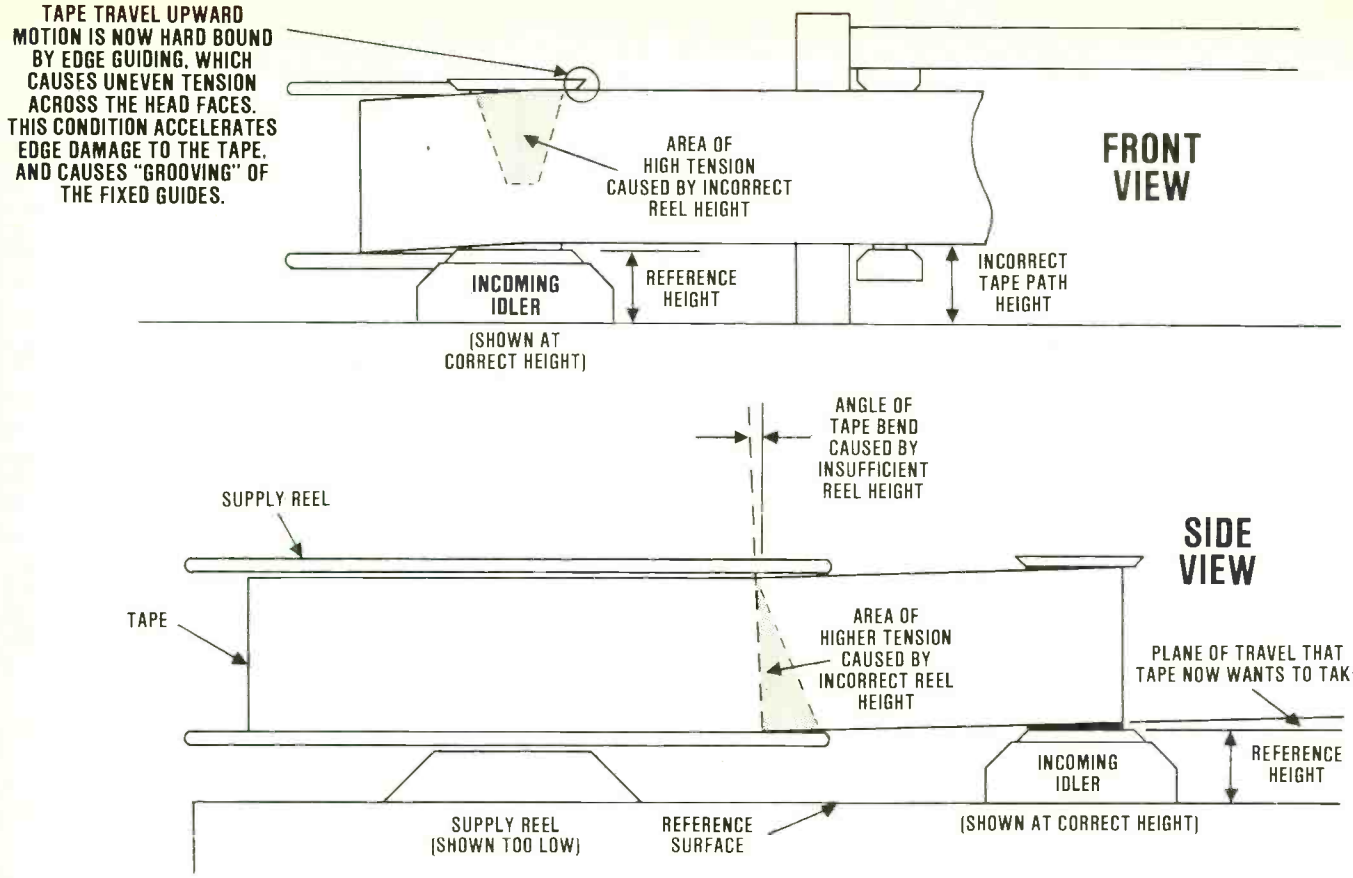

POTENTIAL TAPE TENSION PROBLEMS CAUSED BY INCORRECT SUPPLY-REEL HEIGHT

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coefficient of friction between the roller and the tape is greater than that of the capstan and the tape. As a result, the true tape speed may no longer be equal to the surface speed of the capstan. Because the pressure causes an indentation of the pinch roller, the effective radius of the roller is reduced from R2 to R3. The tape enters and follows the pinch roller during angle $\theta 1$ with speed of V1, and leaves the roller during the angle $\theta 2$.

The surface speed of the pinch roller in the nip area is somewhere between

$$V2 = \omega R2, \text{ and } V3 \leqslant \omega R3,$$

and the tape speed is:

$$V3 \leqslant \text{Tape Speed } V \leqslant V1 < V2.$$

The tape leaves during the angle $\theta 2$ and the surface speed V2 of the pinch roller is greater than the tape speed, V. Therefore the tape will slip and the relative speed difference between V2 and V will be small, so that large scale stick-slip friction occurs. This would also be true if the tape contacted the pinch roller prior to the capstan, which generates a high degree of scrape flutter that is induced into the head area, and is buffered by the capstan in a well designed transport[2].

Pinch rollers are a study in themselves! Over the years, many different types of material, densities, diameters, strike positions, and heights have been tried, in an attempt to minimize the influence of the rotating rubber tyre of the path of the tape. The greatest success has been encountered in cases where the pinch roller is not hard bound to a given zenith, but is allowed to self center on the capstan shaft. It has been noted that the influence of the pinch roller is minimized by grooving the surface of the pinch roller on wider tape widths, to make it seem to the tape that it is being driven by a number of small rollers. On Ampex transports, for example, considerable improvement is made by reducing height of the roller to less than the width of the tape for MM-1000, -1100, and -1200 machines.

The pinch roller contact pressure should be sufficient to maintain intimate contact between the capstan and the tape. The exact figures are given by the manufacturer for the pinch roller pressure and, as can be seen above, this pressure to a great degree will determine the speed accuracy of the transport! The slippage that occurs in the capstan can be minimized by making the capstan diameter as large as possible, a fact that is evidenced in the design of later machines. It is also advisable to retrofit older, low-speed machines (3¾ and 7½ IPS) with a new capstan that has a larger diameter and a lower RPM. The slippage is also exaggerated when the burnished finish of the capstan shaft is polished off by the slip-stick action of the tape. When this occurs, speed errors are greatly increased, and the short-term speed is influenced much more by the supply and take-up reel tensions. (Here you find an explanation for the timing errors that occur on most two-track machines!)

All (or most) of the above discussion has centered upon the open-loop type or transport design, but much of it still holds true for closed-loop designs, such as the 3M-Series of stereo and multitrack machines. With a single capstan, and two pinch rollers engaging different diameters of the single capstan, and two pinch rollers engaging different diameters of the single capstan, as in the 3M transport design, tape tension is maintained across the heads. (With the Technics design, however, the pinch-roller pressure is established as the differential between the incoming and outgoing idlers; the incoming pinch roller contacts the tape on the supply reel side of the capstan, so that the capstan shaft can buffer out the scrape flutter created by stick-slip.) On 3M transports, the incoming idler engages slots in the capstan that should be 0.003 inches smaller in diameter than the outgoing idler engagement position. In other words, the outgoing idler is trying to take more tape out of the loop than the incoming idler is bringing in, and it is this principle that creates the loop tension on such transports.

The most sensitive area of closed-loop design transports is the perpendicularity of the reversing idler on the front

of the transport. If the pinch rollers are adjusted to have too great a tension, then this idler will exhibit a small degree of bending that causes perturbations of the tape path. The tape will try and climb up this idler, while being guided out of the loop at a different height, causing the tape to ride up and down around the reversing idler — a closed condition that should be avoided! Tape speed of a closed-loop machine is also determined by the existence of the burnished surface of the capstan. When this surface becomes shiny after much use, it should be replaced.

It is equally important that tape runs at the proper reference height for both open- and closed-loop type of transport designs. After the tape exits the capstan, it usually encounters either an outgoing idler (which usually comprises some form of counting system) or on the smaller width transports, an end-of-tape arm; the same rules of height and perpendicularity exist here. From this point, the tape encounters the take-up reel. The reel height should also be about 0.003 inches below the reference height, to allow the tape to evenly pack on the reel without being scalloped by the minor variations in reel height due to bent flanges. It should also be noted that the various tape manufacturers use slightly different flange thicknesses. All of which means that a tape machine should be provided with a good straight supply reel that stays with it in the studio, and that the take-up height should be set for the type of tape that the studio uses most often. (To this author at least, it seems a bit impractical to set the reel heights every time a different brand or type of tape is being used.)

Well, that pretty much takes us through the entire transport! Oftentimes, the effects of various parts being in minor misalignment results in the machine being only a bit peevish about maintaining its calibration. But there are other times when the beasty will try and separate every splice that runs through it or, better yet, runs the tape out of the guides, under a flange, and transform two-inch tape to something under ¼-inch! Usually in these situations, the star performer died a month earlier, so duplication of the performance is impossible.

When dealing with the mechanical repair of tape machines, it is always best to be careful and methodical and check out the system from the beginning to the end. Changing a factory adjustment should only be done after you have proven to yourself in two or three different ways that it *is* indeed incorrect. It is with horror that we have encountered machines where the customer, in trying to "correct" the initial mechanical alignment of the machine, has tightened every screw, and removed every shim! When properly performed, it is not uncommon for a complete mechanical overhaul to take two or three days to complete, even when all of the parts needed to complete the job are on hand. The factory is often reticent about providing advice on mechanical alignment because, most of the time, more harm than good comes from unenlightened poking, probing and prodding. Whenever a guide is removed, or a bearing replaced, it is always a good practice to take notes on how many and what size shims were there.

The concept of thinking about *how* and *why* something works the way it does goes a long way whenever service is to be performed. With care and attention, and access to the correct adjustment tools, a tape machine can be kept in perfect tune. This article should prove helpful in showing the way towards keeping the studio hardware running. ■■■

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For general reading and reference to various technical bulletins:
Ampex Technical Support Department, 401 Broadway, Redwood City, CA 94063; (415) 367-2011.
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New 60-input Quad-Eight/Westrex console in TBS dubbing 5/Photo: Kathy Cotter

FILM-SOUND FACILITIES AND SERVICES

A PROGRESS REPORT

by Adrian Zarin

From *The Jazz Singer* onward, motion picture sound has been a field where technological strides often coexist with controversy over the best way to get things done. Present-day Hollywood — with its wealth of scoring, post production, and dubbing facilities — is no exception. New techniques involving computers and videotape are offering expedient and inexpensive alternatives to more traditional methods.

For some time now, videotape, because of the time-saving factor involved, has become more accepted in the film world as a convenient means of handling various aspects of audio post-production. And, increasingly, it is becoming a useful medium for Foley and ADR recordings, plus scoring sessions. For the latter, the transition is being facilitated by computerized systems for generating visual and aural music cues during orchestral scoring dates. But beyond this ancillary role, computer-based musical systems, including but not restricted to the Fairlight CMI and NED Synclavier, in recent years have proven to be a powerful source of film scores. A tour through Hollywood's scoring and dubbing operations — both large and small — reveals video- and computer-based equipment to be high priorities in the acquisition of new technology.

THE BURBANK STUDIOS: EXPANDING DUBBING AND SCORING FACILITIES

The Burbank Studios, jointly owned by Warner Bros. and Columbia Pictures, but also is set up to serve independent producers, provides services for a variety of major film companies, including The Ladd Company, Lorimar, Clint Eastwood's Malpaso Productions, and even Walt Disney Productions. As will be seen later, the facility recently has seen a series of equipment upgrades, including a new Quad Eight/Westrex re-recording console, and a Mitsubishi X-800 digital multitrack.

The TBS Post-Production Sound department, headed by Tom McCormack, includes two scoring and five dubbing stages, along with ADR and Foley studios and extensive optical and mag transfer facilities.

TBS prides itself on being one of the first film studios to become involved with the use of multitrack recording of motion picture and television scores, while simultaneously recording on conventional three-track 35mm mag — a procedure that first began in the early Seventies. The facility utilized specially designed Ampex MM-1000 multitracks equipped with Filmlock, a system designed to frame lock a 16-track with a 35mm mag recorder and projector. Later, this procedure gave way to the use of 24-track tape machines for added flexibility during overdubbing and remixing. By 1977, when the first 32-digital soundtrack recording was made for Disney's *The Black Hole*, it became obvious to TBS staff that a new custom console would need to be designed to accommodate the variety of recording formats which were either already in use, or looming on the horizon.

Such a console was designed by TBS and built by Quad-Eight Electronics (now Quad Eight/Westrex). It is described as the only film-scoring console to be custom-designed from the ground up specifically for 32-track digital recording, as well as all the other tape and film formats. Having been put into service in 1982, the console was used for the digital recording of motion picture scores for *Star Trek II: The Wrath Of Khan* and *The Twilight Zone Movie*.

The scoring console is equipped with 55 inputs, and features 44-frequency equalization with variable bandwidth, as well as high- and lowpass filters on each input. A 40-position monitor section is provided for monitoring two 32-or 24-track machines, as well as six tracks of echo return for the film and stereo mixes, six cue lines, plus dialog and click/effects track. The board can be set up to simultaneously record three-, four-,

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or six-track 35mm film, along with three-track, half-inch tape, mono, 32-track digital, 24-track analog, and two-track stereo.

The recording of the movie score for *Twilight Zone* was considered unique in the number of machines and mixing formats running simultaneously with minimal patching. Two, four-track Dolby-encoded machines were run in tandem with a 32-track Mitsubishi X-800 digital machine, recording on 30 tracks (four-track film mix, individual orchestra section, two-track stereo mix, clicks and SMPTE timecode), a 24-track Ampex MM-1200 analog machine, and two Sony U-Matics with digital PCM processors for the stereo record mix. Playback of any of the above machines required only pressing the appropriate monitor button on the console; no playback patching was required.

The complex' scoring areas consist of two stages: a larger 65- by 80-foot stage capable of accommodating up to 125 musicians; and a smaller 40- by 45-foot stage accommodating 35 players. Recording hardware includes four 24-track machines with Dolby noise reduction, two 35mm film machines on each stage, two Studer A80 four-tracks, and Ampex AG440 two-track and mono machines.

The largest of the five dubbing stages at TBS is Dubbing 5. In the machine room are 17 RCA Model 85S1 reproducers which, with two transport per chassis, translates to 34 individual 35mm sprockets. All the replay machines are switchable for SMPTE or RCA equalization. In addition, the dubbers can accommodate one-, three-, four- or six-track heads. The three mag recorders located in Dubbing 5 are RCA Model 86S1s; like the reproducers, these units are switchable for SMPTE and RCA EQ, and can accommodate one-, three-, four-, and six-track heads.

The 35mm mag reproducers, recorders, and projector are driven from the RCA PX-21 Servolock signal generator, which provides 10-pulse-per-second impulses to lock together all the transports in Dubbing 5. In addition, each mixer — for dialog, effects and music — has his own remote. The PX-21 works hand-in-hand with a Model PA-302 that provides the start cue, stop cue, and counter functions. The start or stop cue footage (feet and frames) are entered at the console by the mixer, the appropriate button (start or stop cue) pushed, and the whole system takes off at six-times replay speed toward the preset footage. When the system gets within 30 feet of the required footage, it slows to normal speed — 24 FPS — until the exact footage point is reached. Another feature is the remote "ADV-RET" system, which allows the mixer to advance or retard the transports individually. All machines assigned to his mix position can be offset by up to 100 frames, a digital readout displaying the offset position of any machine assigned to that remote.

Dubbing 5 also houses arguably one of the largest dubbing consoles in the country. The 24-foot long Quad Eight/Westrex console is really three separate consoles for dialog, effects and music mixing, each with its own record busses, bias control, playback/direct keys, and even power supplies. There are 60 fader inputs with full EQ, and four, 12-by-6 mixers with solo and echo (one each, for dialog and music, and two for effects), adding up to 108 available inputs. Output busses are designated as follows: eight effects, eight dialog, eight music, eight main (composite music/dialog/effects), and music-plus-effects (or minus-dialog) mixes. Any input assigned to either the music of effects sections,; besides coming up on its respective bus and the main bus, also shows up in the Music-plus-Effects bus, which allows a stereo M&E mix to be made at the same time the domestic product is being made, but lacking only the dialog track.

"The console was also designed so that individual tracks from any given section can be assigned to another section at the push of a button," sound head

Kathy Cotter

Head of TBS Post-Production Sound Department, Tom McCormack, in Dubbing 5's machine room, which houses RCA PM-85 and -86 mag dubbers and recorders, plus portable Dolby M16H noise-reduction rack.

The control room (left) and recording area of The Burbank Studios' ADR (Automated Dialog Replacement) and Foley Stage, with engineer Bruce Wright at the controls.

Kathy Cotter

Tom McCormack adds. "Let's say you're handling a lot of dialog, or a lot of effects, and all of a sudden you run out of tracks. You can push a button and send some of those tracks over to another section. The mixing engineer for that section will pick them up on one of his pots, and handle them from there. It's a *lot* easier to do this at the push of a button, than to get up and patch something from one section to the next."

Other custom features on the Dubbing 5 console include 360-degree panpots, which TBS also designed in conjunction with Quad Eight/Westrex. "As opposed to the conventional four-channel panpot, which would be left, right, center and surround with holes in the middle," McCormack explains, "these are actually eight-channel panpots."

As for the other dubbing stages at TBS, Dubbing 2 and Dubbing 3 are equipped with custom 54-input Quad Eight boards that are about two years old. Both consoles are similar to the Dubbing 5 board in most respects, except that they have four overall echo sends rather than 12 discrete ones. Also, they do not have the assign capabilities of the larger board, and lack individual master recording capabilities for each section.

Dubbing 1 has recently undergone a complete overhaul and re-opened in January of this year. Dubbing 4 is still being reconstructed. In terms of consoles, both rooms will be equipped similarly to the other dubbing stages. In addition, the rooms have been acoustically tuned by Jeff Cooper, and will be using JBL 4675 monitors, as opposed to the Altec Voice of the Theatre monitors the facility traditionally has used. Both Dubbing stages 1 and 4 are also slated to be equipped with full video sweetening facilities in the near future.

Plans are also underway to equip both scoring stages at TBS with facilities for scoring to video picture. "We've purchased some video equipment and intend to have full video capabilities on both stages, using the [Audio Kinetics] Q.Lock system to synchronize all of our machines." McCormack continues. "That's the way we see things going. We will have overhead video projectors on our scoring stages, and as part of the video sweetening facilities in our dubbing rooms. You can project an 8-by-10 or 10-foot square picture, which is quite sufficient for most applications. We have utilized 19-inch video monitors in the past, but we're slowly getting into full video capacities."

MUSIC DESIGN GROUP: COMPUTER-ASSISTED VIDEO SCORING SYSTEM

Roy Prendergast, president of Music Design Group, is a staunch advocate of videotape and its role in film-sound recording. Founded some two years ago, MDG comprises a music editorial facility that handles both film and television work, including many made-for-TV

Music Design Group music editor Merelyn Davis preparing a half-inch four-track master for audio-video sweetening.

movies for ABC's Circle Films. Prendergast and his editors come from a film music background, but quickly became familiar with video techniques in developing the company.

"Video has a reputation for being fast and dirty," Prendergast notes. "Certainly, the time and money it saves in getting the job done is its primary advantage in our area. But my contention is that — outside of the picture area — you can do anything on videotape that you can do on film, and you can do it *just* as well. You just have to ask the medium to do it. Traditionally, producers haven't asked video to give them the quality that a film producer demands and expects."

Prendergast has put his contention into practice by developing VideoScore, a complete system for computer-assisted music scoring to videotape. The system was designed to take video scoring and, with a few modifications, Foley and ADR a few steps beyond just working to a click track. It provides complete computer-generated timing notes for the composer and, most uniquely, can superimpose onto a video picture the visual cues — streamers and punches —that film-score conductors and composers have relied on for years. As many *R-e/p* readers will already be aware, in conventional film scoring the work print will be treated with punched holes to show tempo changes and beat counts, while marks or streamers placed diagonally along the film, during subsequent projection will show up as a vertical line moving left to right, and which provides a very handy visual cue to the conductor signalling the arrival of a musical event in the movie's action.

"My approach to the system from the first was to make it resemble the film system as *closely* as possible," he says. "A prototype system was designed by Bill Dietrich, and the software by Jim McCann [who also had worked on the development of Neiman-Tillar ACCESS system — *Ed.*] They had had some experience in video film systems before, which is why I approached them on this project.

"I told them right from the outset: 'This has got to look *just* like film' — I knew the psychology of film producers and composers. I didn't want to invent a whole new cueing system for them to learn, because I knew they'd be resisting it to begin with. I wanted them to come in and see that everything is the *same* as scoring to film — the streams are there, the punches are there; the only difference is you're dealing with a *video* image instead of a *film* image.

"There are faster ways to do it and there are even more accurate ways to do it. But, for the time being, I said, let's keep it just like film. And that's proven to be, I think, a very correct attitude."

On a film project, VideoScore begins its work right after the spotting session. Specific details regarding where music will be placed in the picture is fed into the computer via an alphanumeric keyboard as SMPTE or feet/frames information, and the computer then prints out a detailed set of breakdown notes that the composer uses in writing the score. The notes include precise timeings (to the hundredth of a second), detailed descriptions of the action that accompanies each musical cue, along

MDG's second-generation VideoScore, a computer-assisted music scoring system for use with videotape. Electronically generated streamers and punches are superimposed on the video picture at timecode or feet/frame locations.

Jeff Di Perna

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SHOWSCAN 70mm High-Speed Projection Combined with Interlock Six-Track Film Sound

A Special Report by Larry Blake

In his 1983 movie *Brainstorm*, director/special effects wizard Douglas Trumbull featured a "mind experience" device that could record and transmit reality. While the device obviously was a figment of the scriptwriter's imagination, Trumbull has now developed a motion-picture process called Showscan that almost provides film directors with the picture and sound capturing abilities of *Brainstorm*'s fictional equipment.

In a nutshell, Showscan employs 70mm film photographed and projected at 60 frames per second, with sound reproduced by an electronically interlocked 24 FPS, six-track 35mm mag dubber. The 60 FPS projection rate was arrived at following extensive testing in which material had been photographed and projected at 24, 36, 48, 60, 72, 84 and 96 frames per second. By analyzing electrocardiograms, electro-encephalograms, and galvanic skin response tests of test subjects, it was discovered that 60 FPS produced a much higher response than the slower speeds, and not that much less than the 72, 84 and 96 FPS rates.

To give the maximum number of patrons a sense of participation with the wide (2.20:1 aspect ratio) screen, Showscan theaters employ a basically square shape, with five, 20-seat rows (see accompanying diagram). "Continental"-style seating is used, with aisles only along the side walls. Because of the room's shape, the throw from the projector to the screen is equal to the screen width, whereas in commercial theaters the screen-to-last-seat distance is often three times the screen width. Thus, the floor-to-ceiling, wall-to-wall Showscan screen encompasses the entire field of view even from the last row.

Such a close inspection of the image is supported by the ultra-high resolution of the 65mm negative, whose area is three times that of a 35mm anamorphic image. This is not to mention the fact that, because of the higher frame rate, 2.5 times more information is recorded per second. Combining the high-speed 70mm film with custom projectors, lenses, and lamphouses provides an image many magnitudes brighter and sharper than that obtained by a 70mm blow-up from a 24 FPS negative, which is the best that standard theatrical exhibition has to offer today. Indeed, the beginning of the Showscan short, *New Magic*, might be the first time in film history that the audience is not sure if they are seeing projected film or reality itself.

Theater Sound System

Equal attention has been given to the soundtrack and its theater system, which was designed by Showscan's Evans Wetmore. The six tracks are distributed as left-center-right behind the screen, left and right side walls, and subwoofers (lowpassed at 100 Hz) located in the rear wall.

with all the necessary cue numbers, tempo numbers, etc.

"The computer comes in handy," Prendergast notes, "if the picture editor makes a change after we've already generated the breakdowns. Many times, the changes will fall right in the middle of a musical cue, which means that you have to change the timing on every subsequent cue to reflect the add or delete that the picture editor has made. When you're working in traditional film technique, the editor has to do a good 15 to 20 minutes of mathematics to calculate the changes, and then the breakdown notes have to be retyped. It takes the computer a minute or two to make all the calculations [the system works in either feet and frames or SMPTE timecode] and reprint the breakdown notes."

When the time comes to do the actual scoring session, the VideoScore processor — a very modest-sized package — and CRT are brought to the scoring stage and tied into the video system there. "We always generate backup videocassettes," says Prendergast, "with the video image, timecode, click track and streams and punches recorded right onto the cassette. In the event that the computer decides to do something crazy — which it never has — we don't have to stop the whole session."

Under normal circumstances, the streams and punches are not actually recorded onto videotape with the picture, as Prendergast explains: "What we do is take the signal out of the VCR, regenerate the entire video signal, and at that point the marks are added to the video picture itself. When you consider what can be done now with special effects in video, it's no big deal at all to have a white line move from left to right over a certain period of time, and end on a certain frame."

The system offers quite a bit more flexibility over conventional streams and punches, which are manually added to the film stock — after locating the correct spot on a Moviola — by scraping the emulsion off the film in the appropriate area. While they can be covered over with tape, which also shows up on the screen, film streams and punches are essentially permanent.

This is not the case with the VideoScore streams and punches, however, since they can be deleted or moved around if the composer wishes to change the music timing, or taken out during playback to give an idea of how the final picture and music work together without the visual cues. Because there is no physical handling of film involved, all of these changes are acheived in seconds, leaving the composer more creative time and latitude during the scoring session.

As this article was being prepared, Music Design Group was completing a second generation VideoScore system. Made up of modified Sony hardware and custom software, the latter written by programmer Marc Daniels, the new

system will include several features not found on the first generation. This includes the ability to handle both American and European video frame rates (30 and 25 FPS respectively), 16 and 35mm film formats, 24 and 25 FPS film rates, and all SMPTE timecode formats.

The new VideoScore system will be able to generate streams and punches in 16 different colors, giving composers the ability to color-code cues for their own reference. A new set of services called Composer Aids also will be added, and includes a footage/timecode calculator. The composer will be able to type in any given timecode number and get back the corresponding feet and frame numbers, and *vice versa*.

The Composer Aids service also will help the composer find the ideal click rate for every piece of music. The composer simply enters all of the hits (sync points) he wants to make in a given segment, and the computer will display the click rate that enables him to catch the optimal number of these hits on the downbeat. It will then calculate the bar breakdowns for that particular click rate, and print out a blank four- to six-stave sketch page for the composer to write on.

To make hits that do not fall naturally on the downbeat, VideoScore can subtly vary the speed of the click track. "We give the computer an in-time and an out-time, and tell it we want to move from tempo X to tempo Y," Prendergast explains. "All the computer does is calculate the smoothest possible accelerando or ritard and generate the appropriate click. Traditionally, this is something composers have never asked for, except on big-budget features, because it takes an editor a lot of hours to manually construct a click track with a variable tempo. Each click has to be manually punched into the film, calculating the correct distance between clicks by eye. With the computer, it just take a couple of seconds."

Subsequent marketing of the latest generation VideoScore system will be handled by a new company, Audio Intervisual Design, Los Angeles, which has developed the necessary hardware, including a Sony microcomputer, VCRs, video monitor, and large-screen projectors, plus proprietary interfaces for linking SMPTE timecode with computer data. A complete VideoScore system is expected to cost in the region of $25,000, a significant cost saving compared to the $150,000 needed to set up a film-based system. In addition, AID will be responsible for formulating customized facilities, and for supervising system installation.

The function of VideoScore ends with the scoring session. The Music Design Group, however, provides complete editing services. "Everything in-house is handled on tape," says Prendergast. "Even if it's a film project, we assemble our units on tape and transfer that back to film."

... *continued on page 88* —

SHOWSCAN 70mm with Six-Track Interlock — *continued* ...

The five, full-frequency main tracks are Dolby encoded on the 35mm mag, and their respective speaker channels (Renkus-Heinz FRS-1582-CB) are voiced using Dolby Cat. No. 64 equalization cards. The constant-directivity high-frequency horns are mounted above the screen to eliminate through-the-screen HF loss.

BGW 750-B amplifiers power both the left-center-right screen speakers, and the subwoofers (Cerwin-Vega L 36-JE) in the rear wall. The left and right surround speakers are currently fed by QSC 3350 amplifiers.

Sound editing and re-recording for Showscan films are done on standard 24 FPS, 35mm mag using a 24 FPS anamorphic reduction print skip-printed from the 60 FPS 65mm negative. Since some of the 65mm frames don't make the cut to the editing workprint, sync often has to be checked by running a cut track in interlock with the 70mm 60 FPS contact print. The 24 FPS, 35mm anamorphic workprint is also used to make the ¾-inch U-Matic videocassette dubs for ADR and Foley sessions at Lion's Gate Films in West Los Angeles, and for use by composer Herb Pilhofer in his studio in Minneapolis.

Pilhofer, who has composed the music for all Showscan films, makes use of a studio featuring an Audio Kinetic Q.Lock 3.10 SMPTE synchronizer, and dual Otari MTR-90 24-tracks in conjunction with an automated MCI JH-600 console. The multitrack music tapes were mixed down to SMPTE-coded one-inch eight-track tape, the six tracks of audio information corresponding to the loudspeaker configuration in Showscan theaters. This second-generation tape was locked to the film chain during the final dubbing at Lion's Gate.

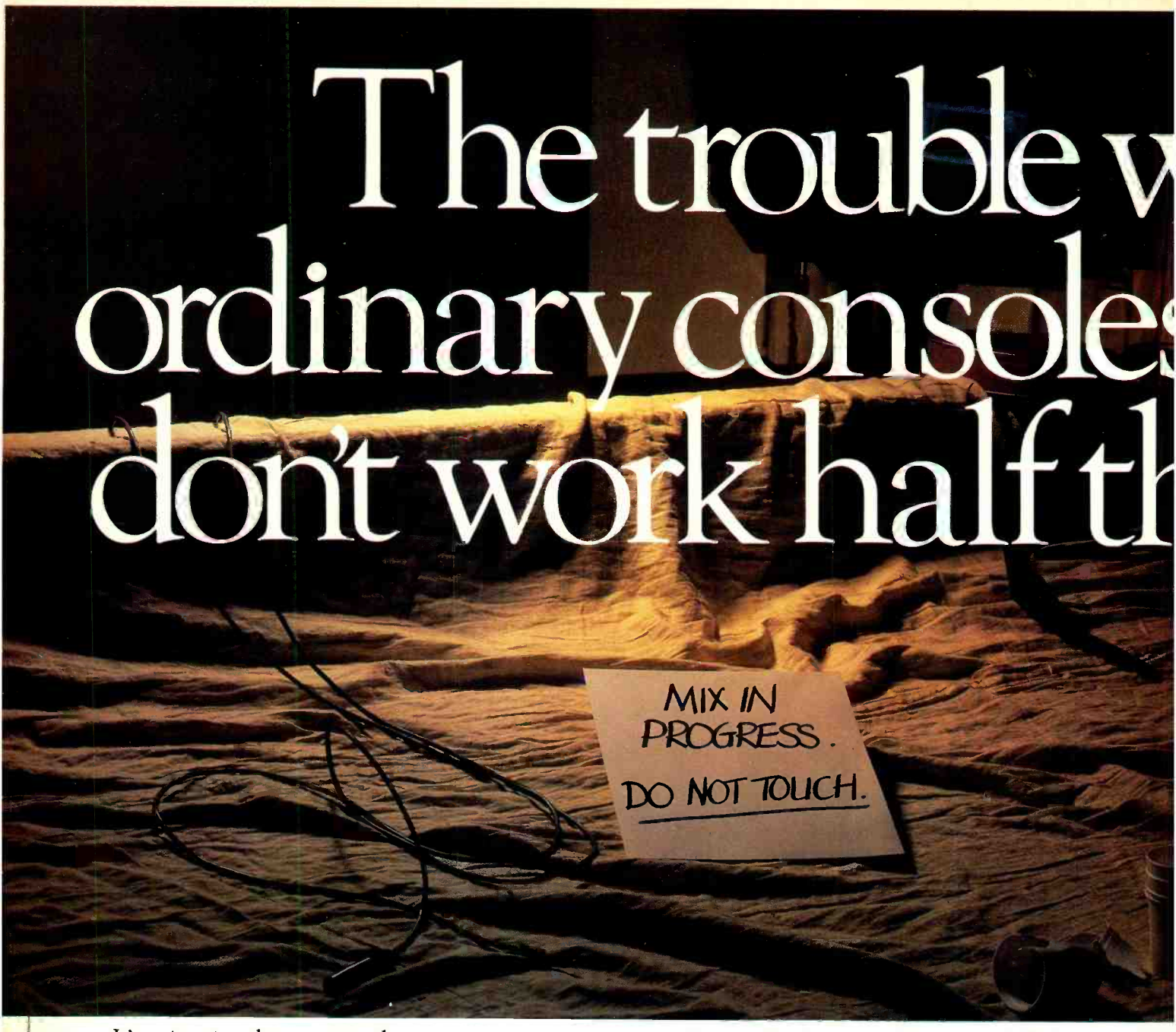
Initial Showscan Installations

Looking for a partner to help with the distribution of Showscan films, in 1982 Douglas Trumbull approached Bob Brock, president of Brock Hotel Corporation, the parent company of ShowBiz Pizza Place. The ShowBiz concept included, in addition to a pizza parlor, a complete family entertainment center, and the idea was to build the special Showscan theaters in existing ShowBiz locations, as opposed to converting standard theaters. With this in mind, the standard Showscan theater is designed to fit easily into a structure with only 17 foot clearance.

Initially, all of the Showscan theaters will be located in ShowBiz Pizza Place entertainment centers, which are to be found in shopping centers throughout the United States. The first four theaters opened in February 1984 in Huntsville, Alabama, Fairfax, Virginia, Springfield, Missouri, and Dallas. Current plans include expansion to 150 markets by the Fall.

The first two Showscan shorts to be seen in these theaters, *New Magic* and *Big Ball*, are 22 minutes in length, allowing for showings every half-hour. Admission prices are currently $2. In the future, Trumbull would like to see the development of longer films and serialized "cliff-hangers."

The trouble w ordinary consoles don't work half th



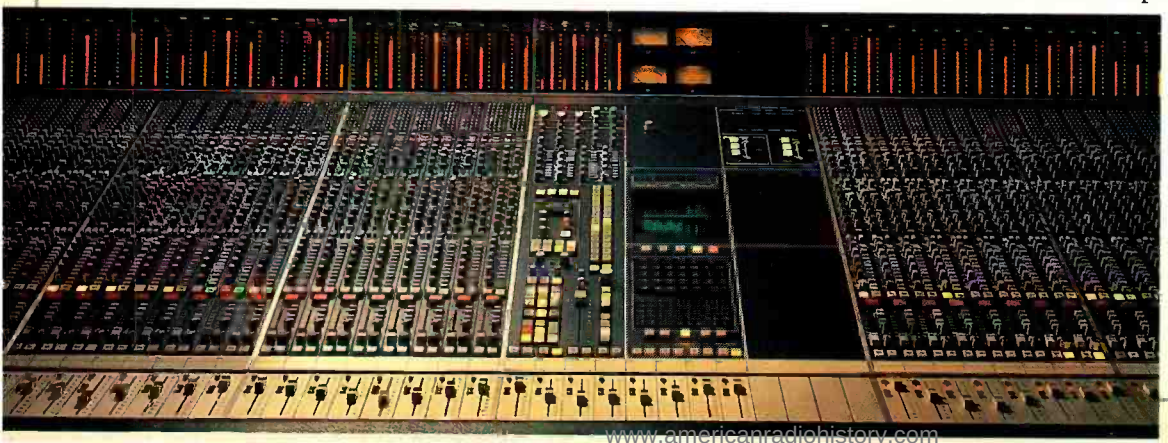
It's a situation that every studio manager recognises. A client has been in, done some work, and departed to return some time later. Expecting to find the desk as it was left.

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

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

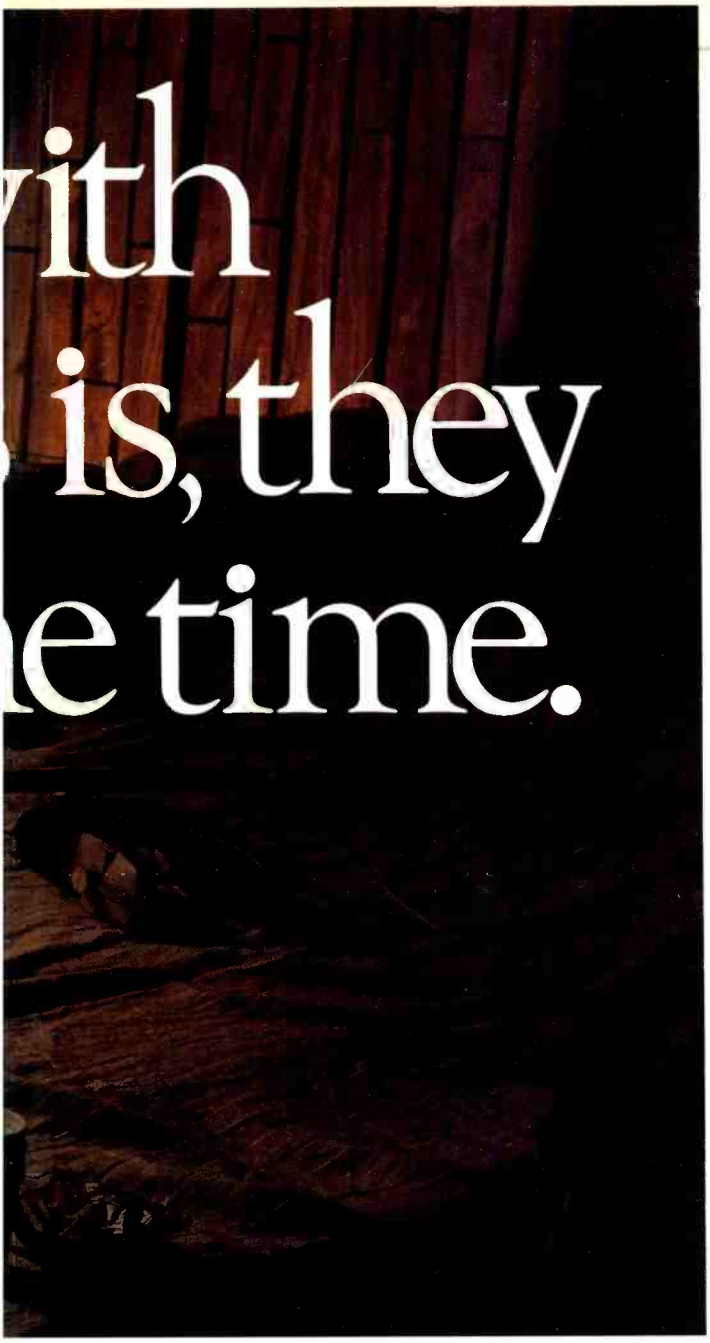
At Solid State Logic, however, we've developed a rather more practical solution to this dilemma. We call it the Total Recall System.

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



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

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



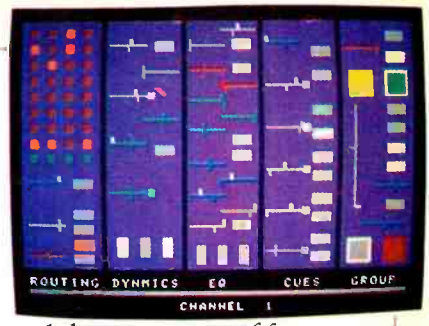
to save time at a later re-mix. And engineers can even store their personal EQ and dynamics settings and create their own libraries on floppy disc.

Total Recall is just one of the functions of the SL 4000 E's on-board computer. The computer will record all the details of a session – title entries, track lists, cue points, dynamic mixes, synchroniser information and so on – and store them on a floppy disc.

These unique facilities give the Solid State Logic Master Studio System several important advantages.

It allows the manager to keep his studio working, and earning, for the full 24 hours a day. Because even the most complex set-up can be precisely reproduced in about 20 minutes before the start of a session.

It saves the engineer wasting precious time and lets him concentrate on the creative process, from track laying to over-dubbing through to mixing. Because the studio computer speeds up everyday tasks like autolocation, drop-ins, mixing and synchronisation.



And it gives producers and musicians real flexibility and continuity. After recording in an SSL studio, they can return there (or to any other computerised SSL studio in the world) and continue work with absolute accuracy and the minimum of fuss.

Yet the computer is simplicity itself to operate. Even inexperienced assistants and tape-operators will soon master its basic functions. While feed-back from studios with SSL systems shows that more advanced expertise is acquired quickly and naturally with use.

The SL 4000 E Series Master Studio System could only have been developed through an understanding of the needs and problems of people who spend their lives in studios.

So it's not surprising that SSL's design team include not only computer and electronics experts, but engineers, producers and musicians. People who both improve studio technology and use it.

What this group sets out to devise, and SSL sets out to produce, are real answers to real problems. Finding ways to improve quality and streamline audio production. But we also produce machines that are built to last.

You will never outgrow an SL 4000 E System because you can start with the basic mainframe and then add extra modules and facilities as your needs and budget dictate. By designing systems with the future in mind we make sure that hardware and software developments can always be integrated into existing systems.

We can build you an SL 4000 E Series Master Studio System in around three months. So if you would like to start cutting the amount of time and money your studio wastes, cut the coupon or call Antony David in the UK, Doug Dickey or Piers Plaskitt in the USA.

Solid State Logic

Please send me further information on the SL 4000 E Series Master Studio System.

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Tel: (099) 389 8282. Telex 837400 SSL OX G. Facsimile (099) 389 8227.

Solid State Logic Inc., P.O. Box 200 Milan, Michigan, 48160, USA.
Tel: (313) 439 8866. Telex 230504. SSL MLAN. Facsimile (313) 439 8516.

SHOWSCAN 70mm with Six-Track Interlock — *continued . . .*

Also an obvious contender for future Showscan presentation is the performing arts, documenting great dance, opera, stage, and musical performances. In addition, one can look forward to Showscan films laden with visual effects, like those created by Trumbull for *Close Encounters of the Third Kind*, *Blade Runner*, and *Star Trek: The Motion Picture*.

So far, no dramatic sync-sound films have been made, and the loud noise of the high-speed 65mm camera (moving the film at 56.1 IPS!) precludes any close-up dialog recording. Sound editor Sandy Berman reports that in some scenes of *Big Ball* the camera noise was so loud that they had to refer to the script to know what the actor was actually saying!

Historical Background

As the old saying goes, nothing is new except that which has been forgotten, and Douglas Trumbull has, with Showscan, given a new twist to equipment, techniques, and principles explored by previous wide-screen processes.

First and foremost is Cinerama, which is the grandaddy of all wide-screen, stereo-sound formats. The most notable similarity concerns the fact that Showscan continues and improves upon Cinerama's intent of simulating reality by totally encompassing the field of view of the audience. One of the problems of Cinerama was that the screen was so deeply curved (146 degrees) that the theater's "sweet spot" was very small. In addition, the high installation costs of the three-projector, 75-foot-screen Cinerama system also limited the number of theaters — probably no more than 30 Cinerama theaters were in use at one time in the US.

Showscan neatly sidesteps these problems by being *designed* to be shown in small 60- to 100-seat theaters with a large-appearing screen, although larger theaters could be used and are on the drawing boards. Not only is the installation cost minimized in these small theaters, but the standardized slight curve of the screen and the seats ensures that all patrons have an unobstructed view and complete involvement.

The sound for Cinerama films was, as with Showscan, reproduced in double-system interlock using 35mm fullcoat. In fact, it is almost axiomatic that any improvement in film sound involves a soundtrack that is *not* married with the picture print. Vitaphone, Fantasound (*Fantasia*), and Cinerama are quintessential examples in this regard, and the chances are good that the first digital motion-picture sound presentation will keep this streak going.

The second format in whose footsteps Showscan follows is Todd-AO, which standardized the 70mm process in use today. (Wider-than-35mm films have been experimented with since the beginning of the century.) The first two Todd-AO films, *Oklahoma!* and *Around the World in 80 Days*, were photographed at 30 FPS, primarily to reduce strobing on the wide screen. However,

— *continued from page 85 . . .*

The facility has developed a system of electronic editing using a BTX Shadow SMPTE synchronizer. "What we basically do is pre-assemble edit from one [four-track] half-inch tape machine onto another," Prendergast explains, by sequentially recording or dropping-in on a slave while replaying tapes from the master against timecode. "One of the [Otari MTR-10] machines has timecode on it matching the timecode on the picture. Because we don't have to razor-blade edit, the timecode remains intact and we are able even with videotape to deliver the music in sync with the picture, just as one ordinarily does on a film project."

On the whole, Prendergast feels confident that, thanks to computer assistance, videotape is on the way toward gaining the same kind of acceptance on the scoring stage that it already enjoys in other areas of the film chain. "About the only objection we still run into is the size of the video picture," he notes. "Film people like that big screen. Even this, though, is starting to fall by the wayside. On the last picture we did, the producer was absolutely delighted with the new Sony eight-foot projection screen that we used on that session. Apart from the size factor, he was able to work with a clean, color picture, as opposed to the dirty black and white work print that is the norm in film scoring. His comment was, 'This is the way I want to do *all* my shows'!"

MUSYNC: SCORING TO PICTURE . . . WITH A DIFFERENCE

Designed by film composer Robert Randles, Musync represents another form of computer-assisted scoring. It has many of the same specific capabilities as MDG's VideoScore system, but was designed on a different premise. While VideoScore offers all the trappings of traditional film scoring on videotape, Musync is set up to work with either videotape or film, and is *not* dedicated to imitating film streamers and punches.

"If you want streams and punches, you can have them; but you're not utilizing the capabilities of the system to their full extent," says Randles. "What we have with streams and punches is a very crude way of handling graphic display, and our capabilities have been expanded tremendously. Why design everything around something that happened because that's all we could do at the time?"

Randles evolved Musync over the past nine years, using a combination of stock computer components and custom hardware. "The procedure of how you use Musync varies as much as the personal styles of different composers," he states. "You decide the format that you're working in. Musync will count film footage or SMPTE timecode, but that's for it to worry about, and not you.

Photo courtesy of AMPAS

Bob Randles demonstrating Musync to members of the Academy of Motion Pictures Arts and Sciences.

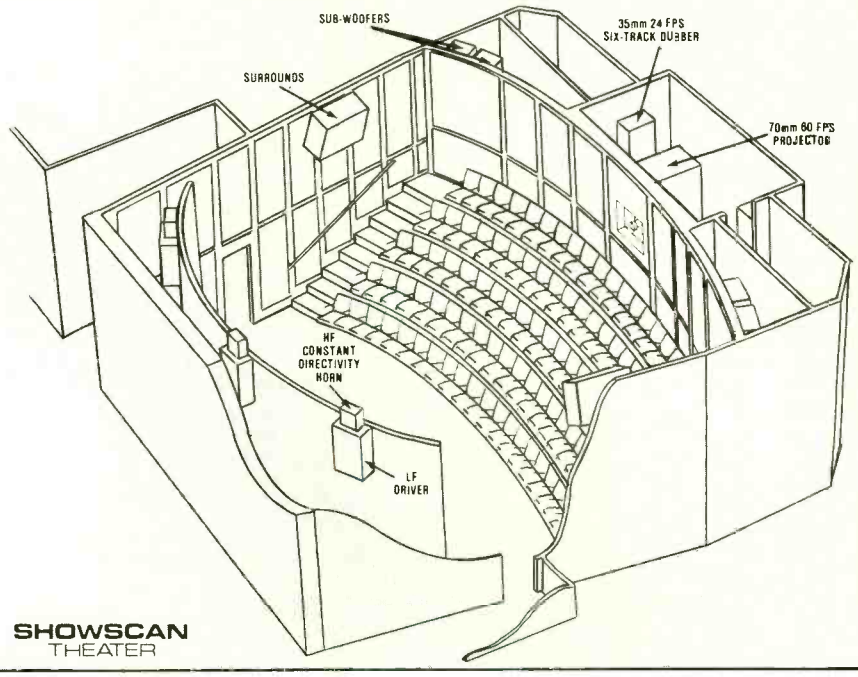
"The system came about basically because machines measure film and music one way — in arbitrarily spaced units of time or physical length — and people measure them in another way, in terms of subjective events. Musync lets the composer forget about the numbers, and think only in *human* terms."

After the spotting session, the music editor goes through the picture, typing verbal descriptions into Musync at each point the editor believes is significant to the composer. Musync automatically assigns a label and associates the description with the frame that was in the gate or the timecode number on the head when the typing began. If the music editor doesn't feel like verbalizing, Musync can just automatically assign a letter name to the spot as well. Aside from

SHOWSCAN 70mm with Six-Track Interlock — *continued . . .*

because a 35mm release would take place after the initial 70mm "roadshow" engagements, a second camera was used on both films to record a 24 FPS version. All later 70mm films (including those shot in the Ultra- and Super-Panavision processes) were photographed at 24 FPS.

Although the sound for the first few engagements of *Oklahoma!* was reproduced in double-system six-track interlock, as in Showscan, later films have employed striped prints. Tests have shown that the striping is sheared off in high-speed Showscan projection, and for the time being the high quality of 35mm magnetic film will be exploited, despite the potential sync problem that is present with any double-system screening. □□□



this, it also handles events of duration, such as dialog, and graphically represents overlapping events."

The system generates a CRT display of this information in juxtaposition with musical beats for planning and editing musical tempo. It also prints out a bar sheet. "Years ago, we used bar sheets more than we do now," Randles notes. "The music editor in the past would write out the visual events on the bars corresponding to the music that accompanies those events. That was a *very* time-consuming and labor-intensive process, but Musync does it very quickly."

Depending on the format used, and facilities available at the scoring stage, the Musync prompter display can be shown in a variety of ways during the scoring session. "You can take movie film, project it on a screen, and project the prompter display on top of that if you want," Randles says. "Or, you could have the prompter display set up at the conductor's podium just for him to see, and have the film on a larger screen. The prompter display is easy to follow, and this setup is usually quite sufficient."

The display includes a sort of visual click track that accompanies the aural click track also generated by Musync. Randles describes it as "sort of a bouncing ball idea which augments and makes practical for the first time a variable click track. It also makes the click track optional. Because the display is so easy to follow, you can turn off the clicks and follow precisely. Or, you can be freer with your tempo, and get back on beat when you need to."

Musync's variable-click capability and visual counterpart came in handy, according to Randles, in the course of his work on the score for *Rumblefish*. Since he was inexperienced in film scoring, Police drummer Stewart Copeland provided ample opportunity for Randles to test Musync's ability to handle unconventional demands, and at the same time to observe some of the ways in which traditional scoring techniques have imposed creative limitations on the composer in the past.

"For the scene that signals the dramatic appearance of Rusty James' brother, The Motorcycle Boy, Stewart improvised a radically rubato passage on the synthesizer," Randles reports. "He told me he wanted the French horns to play along with the part, overdubbing and exactly matching the beat. Of course, it would be impractical, if not downright impossible, to do this using conventional methods. Under ordinary circumstances, a composer asking for something like that would probably be considered crazy.

"The French horn players *were* able to do it, however, using the Musync prompter display. They could see the next beat coming; anticipate when it was going to happen — as opposed to sitting there waiting for the next click . . . waiting for the next shoe to fall as it were! They were able to overdub the passage perfectly on what I recall to have been the first take. It precisely matched the original track, but came out sounding very free-flowing and uncontrived."

The *Rumblefish* score marks the most extensive use of the Musync system to date, but Randles has used it on a number of other projects, including *Conan the Barbarian, The Outsiders,* teasers and trailers for *Outland, Blade Runner,* and *E.T.*, a Universal Pictures log, and various other commercials and trailers. He looks upon Musync as being in a continual process of development. Currently, he's looking into applications of the system in editing, ADR, and Foley work, and the possibilities of interfacing with synthesizers, sequencers, and related equipment.

HK SOUND: CREATIVE USE OF SYNTHESIZERS IN FILM SCORES

HK Sound is one of several companies currently exploring the advantages that computer synthesizer equipment — in this case, the Fairlight CMI — offers to scoring. The company is headed by art director and computer-graphics expert Dale Herigstad, and composer/programmer Jerry Kaywell. The pair use a Fairlight for a variety of projects, ranging from TV ads to full-length films.

Thanks to the Fairlight's capabilities as a multitrack sequencer, Kaywell is able to work with a relatively modest complement of recording equipment, located in his one-room home studio. Control-room hardware includes a modified Tascam 80-8 eight-track on half-inch recorder, StudioMaster 16-by-8 and 8-by-4 mixers, and an extensive custom patchbay with 144 patch points. Combined with the Fairlight's eight separate outputs, this set-up provides Kaywell with all of the tracks he needs.

Output from the Fairlight is first routed to the 8-by-4 submixer, and then to the main mixer. "I'll usually assign all of the Fairlight tracks — all eight of those output streams — to one channel of the [16/8] mixer," Kaywell explains. "I'll just keep working on the Fairlight tracks until they're mixed exactly the way I want, and then print them on just one track of tape.

"The patchbay gives me EQ at various points in the signal path, so I can tailor the sound quite nicely. For this reason, I always go for a maximum signal-to-noise ratio on tape. With the use of a sync tone, you can dump eight Fairlight tracks on to each track of the tape machine [in multiple passes], and get up to 64 [synchronized] tracks."

Kaywell masters to a TEAC quarter-inch two-track. For mono television productions, he places a 60 Hz sync pulse on the right channel, and music on the left. Stereo masters and mag transfers are handled out of house.

At the heart of the Fairlight's power as a computer-assisted scoring device is its Music Composition Language. Kaywell has developed his own system for referencing MCL to film feet and frames. "I can assign a whole note a value of 96 microbeats on the system's speed clock," he explains. "Ninety six is divisible by 24, and film runs at 24 frames-per-second. If a whole note equals one second, or 24 frames, a half note will have a value of 48 microbeats, and equal half a second [12 frames]; a quarter note is six frames, etc.

"When I get a piece of film to be scored, I start by taking notes and making timing sheets. Let's say a door opens

The Musync computer-assisted scoring system consists of (left to right) a Prompter Display, Viewing Bench, and the Working Display Unit used by the music editor and composer to "spot" the film for music cues, and provide annotated comments for the bar sheets.

Composer Jerry Kaywell developing custom sounds at HK Sound's Fairlight CMI digital synthesizer.

up at five frames, five seconds, and I want to have music begin as soon as the door opens, play for the amount of time it takes the door to open, and then stop. I'll program the Fairlight to rest for five whole notes — that's five seconds — and for 5/24 of a whole note — that's five frames. Then I'll command it to play a note when the door opens. Say it's a 36-frame door open; I'll just command the Fairlight to play the note for 36/24.

"Then, I can start tailoring the sound. Maybe I don't like the way the sound cuts off at the end of the note — it's too abrupt. I'll start playing with the damping, programming the note to play at full volume for 12 frames instead of 36, and having it progressively decrease in volume at a fixed frame rate."

As with the VideoScore and Musync computer systems, the Fairlight's MCL easily handles tempo and timing changes. Accelerandos and ritards can be constructed to picture requirements; and because the music is actually played by the computer rather than human musicians, it can execute tempo changes and other effects that would be impossible for even the best session players.

The Fairlight also offers the composer the advantage of digital time compression, Kaywall points out. "Sometimes I've done tracks that were supposed to be exactly 30 seconds but which ended up being something like 31½ seconds. In a case like that, all I need to do is type in a speed number and use digital compression to make the music fit. And with none of the funny sound you get from VSOing the tape."

Of course, the entire score can also be adjusted to take into account last-minute picture changes. "That's one of the strengths of this company," says Kaywell. "We an accommodate all sorts of changes at no extra cost. All I do is type in a few commands, and the music cues and effects are brought into line with the revised picture. There's no second call for musicians, and all the time and expense that involves. There was no *first* call, as a matter of fact! All the adjustments are made by one guy at the keyboard."

Whenever possible, Kaywell likes to work to a reference picture as well as timing notes. "I like to get a transfer of the film to videocassette with visible timecode on it," he explains. "I look at the picture, sync up the Fairlight, and I'm actually working to picture."

Kaywell synchronizes the Fairlight to picture without employing a synchronizer. "When I get visible timecode on my reference videocassette," he says, "I always ask to have all zeros until the first frame of the picture, and then the numbers begin with the first frame. That way I can use the 'Wait' function on the Fairlight to synchronize to picture. You type 'Wait = On' [on the CMI's keyboard], which means that the music is going to hold until you depress any note on the keyboard. It then comes on instantaneously. I start the music with the first frame of the picture, and it's right on the money. Since everything is done in-house, I really don't need a synchronizer."

Creatively, the Fairlight offers the composer numerous advantages, according to Kaywell. "For one, I can compose with the final sounds right there — rather than sitting at a piano and envisioning things. I'm actually building the track as I'm composing. Plus, the Fairlight remembers everything for me and will print out the musical score, so I have no need for a copywriter."

HK Sound has a library of over 2,000

Fairlight instruments and sounds, and is constantly adding to the stockpile. Such an extensive library provides the studio with the ability to handle effects as well as music. "If I get a picture that shows ice dropping into a glass in slow motion," explains Kaywell, "I can sample the sound of an ice cube being dropped into a glass, bring it down an octave or so, and get a sound that's a *precise* aural rendering of the picture. It's all done digitally, and the sound can be altered and manipulated at will. So there are none of the compromises that always seem to be involved with conventional sound effects on analog tape."

By generating both the effects and musical score on the Fairlight, Kaywell is able to take a holistic approach to the soundtrack. "A lot of times," he says, "if the piece requires many sound effects, I'll compose them first, and listen to them for direction in writing the music. Do the effects suggest a rhythm to me? Do they suggest the sounds of certain instruments? Then, of course, when the music is written, I can pitch the sound effects in key with the music."

Kaywell also makes great use of the Fairlight's ability to rapidly re-assign voices. For example, a passage written for violins might work better played by bells, and can be changed around to do so in a matter of seconds. As a composer, however, Kaywell tends to shy away from using the Fairlight to imitate violins, pianos, etc., and prefers to exploit the instrument's potential for sounds that have never been heard before.

RCA mag dual dubber/recorders, MCI 24-track, & BTX Shadow synchronizers at MRI.

"Sure, it's very cost effective to have the Fairlight imitate grand pianos, string or horn sections, and things like that," he comments. "But I still say, 'If you want a grand piano sound, go *mike* a grand piano!' The Fairlight is best used for things you couldn't do without it — like getting instruments to play backwards, and then all of a sudden turn around and play frontwards.

"Sound manipulation is what this technology is *all* about, so I'm constantly looking to go in that direction —especially in film. The sophistication of film today lies in its visual trickery; there's *so* much of it that we need some *audio* trickery to match it. The Fairlight to me is a powerhouse of that trickery —and that makes for effective scoring. Combine that [ability] with the fact that you can compose right on the frame, and you have an unbeatable team with the Fairlight and film."

VILLAGE/MRI: A HYBRID "ONE-STOP" FACILITY FOR FILM AND VIDEO

An ideal place to conclude our tour of current developments in film sound services and facilities is with the recently formed alliance between Hollywood's Motionpicture Recording Inc. (MRI) and The Village Recorders, based in West Los Angeles. The two facilities have joined forces to provide a comprehensive package of services ranging from scoring to post-production on both film and videotape. "The concept is to create a one-stop shop," says Michael Geller of the Village. "Clients who would normally record their score here, and then bring the rest of their budgets to other facilities, can now find everything they need under one roof, figuratively speaking. At the end of the project, they pay one check to one facility."

The Motionpicture Recording facility, headed up by Garry Ulmer, centers around a four-track stereo film dubbing stage, which is also equipped to handle film-based ADR work. Serviced by RCA dubbers and an RCA/Photophone projection system, the command post for the dubbing stage is an AMEK console that was custom designed in conjunction with Ulmer.

"It has all the features of a record console, but re-adapted to fit the film configuration," Ulmer notes. "Separate music, dialog, and effects sections were designed, each of which has 12 inputs [a total of 36 inputs], each assignable to any of 24 outputs."

In designing the submastering facilities for the board, flexibility was the key concern, Ulmer explains, "Submastering can be done from any section, grouping any modules together. Most film dubbers have assigned groupers that can only group their own section — either music, dialog, or effects. But I can make any module a group that can [control the overall level of] anything from one to 36 inputs."

The AMEK console also includes onboard graphic EQ and four-band parametric EQ with high- and lowpass filters. Full-range equalization facilities were a priority for Ulmer in designing the console.

"Although mono optical work doesn't accept certain frequencies, and therefore limits you to a specific range, there are other projects that have a broader range and call for the full EQ capacity," he says. "There's Dolby Stereo, for one, which doesn't have unlimited range, but you're still mixing to a flat response, without an Academy rolloff. All of your

MRI's main dubbing room houses a 36-input AMEK console modified to accommodate the music/dialog/effects format needed for film sound re-recording sessions.

television commercials and film trailers for TV are mixed to a flat response, because they go to videotape.

"I feel another advantage of the full range of equalization lies in the Music Videos that are now being produced. They're [usually] being shot on film and, in essence, are becoming concentrated mini-movies. There are effects, dialog and music, all of which can be mixed on a dubbing stage and then go to videotape, where the extended range again will be needed."

MRI's console is automation-ready, according to Ulmer. "It's all VCA faders and groupings, and all the terminals are set for the computer; it's just a matter of hooking it up."

Projects at the MRI dubbing stage are mastered to mag stock: either four-stripe, three-stripe, or single-stripe. The console has the capacity to master on audio tape as well but, according to Ulmer, there is generally no call for this capability.

Across the street from Motionpicture Recording on Hollywood Boulevard lies the newly-constructed Village/MRI studio, a complete video sweetening room with videotape-based ADR and Foley facilities, and complete mono and stereo optical transfer equipment. The room features an automated API console, an Ampex MM-1200 24-track and BTX Softouch synchronizer. Video decks are all ¾-inch Sony U-Matics, with Pulsar projection equipment and a 10-foot screen.

Ulmer acknowledges that the decision whether to handle ADR work on film or videotape involves a number of variables. "It's a toss-up," he concedes. "If the client has the time and is well prepared, videotape is the way to go. If the client is rushed, or just popping lines in here and there, it's probably better to stay on film. It's faster on videotape, but film can be more expedient in some situations. Another consideration is the medium you're ending up on. If you're going to end up on videotape, it pays to do your ADR on tape. If you're ending up on film and have a lot of ADR work to do — hundreds of lines — then I'd still advise going to videotape. But if you have just a few lines and are going to end up on film, it's better to stay on film. Fortunately, we can go either way here."

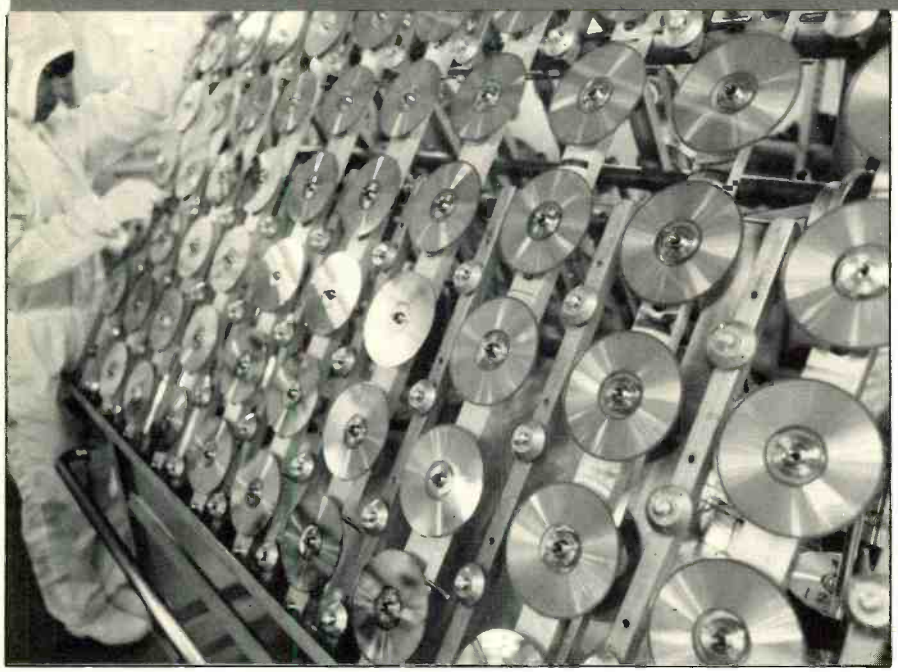
Scoring is handled at the Village facility in West Los Angeles. In recent months, the studio has completed the music for *Yentl* and several other features. Scoring is primarily handled in Studio D (which, like all of the Village's rooms, is Studer/Neve equipped) using videotape and a click track. Studio D is equipped with a newly-installed Sony video projection system with a 100-inch screen.

The Village has recently purchased some digital multitrack recording equipment, which it intends to use in its scoring operation. "We're just at the beginnings of that technology," says Michael Geller. "We're recording some albums digitally and mastered the *Yentl* [film soundtrack album] on digital, but we've yet to do our first score on multitrack digital."

Another aspect of the Village's increasingly thorough commitment to film work and digital audio stems from owner Georgie Hormel's position as North American distributor for Fairlight. "We're preparing to get involved with computerized digital Foley work using the Fairlight," Geller comments. "It's going to be a long, hard road to get people to accept things like this, but we feel there's quite a future in this area."

Currently on the drawing board are two more expansions for the Village, which are also related to film and video work. One is a ½-inch high-band component video edit bay, which will complement their present ¾-inch on-line/off-line video editing facilities. And on the second floor of the Village complex, construction is already underway on a combination video production/large orchestral scoring room.

"The main wall is built already," Geller reports. "It will be a room for large orchestral scoring. Plus there will be lighting grids so the room can double as a video production facility suitable for shooting a rock clip. With the completion of these new facilities, we feel we'll be able to offer our clients a truly comprehensive motion picture/video package." ■■■



COMPACT DISC MASTERING AND MANUFACTURING

A Sneak Preview of the Process to be Used at the New CBS/Sony Terre Haute Facility

by Mel Lambert

Since the late Seventies, when Philips and Sony first began co-development of the Digital Audio Disk, and the subsequent decision to base consumer hardware on the Compact Disc format, there is little doubt that the impact of digital technology on the pro-audio industry has followed closely the genesis and acceptance of CD players and software. And with projected US sales this year of some 150,000 Compact Disc players (compared to 40,000 in calendar 1983), it would appear that the growing number of facilities around the country investing in digital recording equipment are set for a profitable future. (To put the blossoming market for Compact Discs into per spective, Michael Jackson's *Thriller* CD, so far the best-selling international Compact Disc release,to date has shipped in the region of 20,000 copies worldwide.)

Seemingly in response to the increasing demand for CD releases, August of this year will mark the scheduled opening of the first major Compact Disc manufacturing facility to be established in the United States. The new facility, to be based at a converted CBS tape manufacturing plant in Terre Haute, Indiana, will employ 130 people, and has a predicted maximum production capacity of some 11 million CDs per year. (Initially, the facility will produce close to 300,000 CDs per month, building up to a total monthly capacity approaching one million once the plant comes fully on line.)

According to CBS/Sony, the new Terre Haute Compact Disc pressing plant will be modelled closely on the company's existing Japanese facility, located in Shizouka, 125 miles southwest of Tokyo, and which began manufacturing Compact Discs in late 1982. Production at the Shizouka plant totalled 1.2 million Compact Discs in the first three months of this year, and is expected to reach 3.6 million fo 1984. (The Shizouka factory also serves as a "conventional" vinyl pressing plant; 20 million albums, 13 million singles/EPs, and 10 million Compact Cassettes were produced there last year.)

While there are certain operational differences in the way Compact Discs are manufactured at various pressing facilities, the basic process is very similar at the nine existing CD plants around the world: Polygram in Europe, and CBS/Sony, Toshiba/EMI, Denon, Sanyo, Pioneer, Matsushita, Nippon-Columbia, and JVC in Japan. (As Roger Nichols pointed out in his article, "Preparing Digital Master Tapes for Compact Disc Mastering," published in the April '84 issue of *R-e/p*, all nine plants, with the exception of JVC, which specifies DAS-90/900 PCM encoding, require that the master tape be in the form of a Sony PCM-1610-encoded U-Matic videotape; PQ Subcode formats also vary slightly from manufacturer to manufacturer. It should also be noted that several digital editing and post-production facilities around the US are set up to handle direct digital-to-digital dubbing, and can accommodate DAS-90/900, PCM-F1/710, PCM-1610, Soundstream, 3M, and Mitsubishi X-80 format material.)

As I discovered during a recent visit to the Shizouka plant, the Compact Disc manufacturing process can be divided into four distinct stages, identified at the CBS/Sony facility as the A-Room, where the laser-etched glass master is prepared; the B-Room, in which the metal master and stampers are prepared; the C-Room, where the plastic CDs are injection moulded and receive their reflective and protective coatings; and finally the D-Room, in which the labels are printed directly onto the top surface, and the final product is wrapped and packaged. To prevent contamination of the various materials from dust and airborne particles, the first two preparation areas — laser cutting, and master/stamper production — are operated as clean rooms.

A-Room — Laser Cutting Process

Assuming that the master tape is in a format that can be handled by the pressing plant, the digital information and PQ date can then be transferred directly to a special optically-coated glass plate that later is used to prepare metalwork for the CD injection moulding process. To enable continuous preparation of glass masters, the new Terre Haute plant will be equipped with a pair of Sony DMC-1100 Disc Master Code Cutter units. The DMC-1100 comprises a computer-controlled laser cutting "lathe", and an associated rack unit that houses a PQ Code reader, a PCM-1610 digital processor, and companion BVU-800DA U-Matic VCR.

According to the Terre Haute plant's VP/general manager, Jim Frische, the mastering equipment is designed for virtually automatic operation; once the U-Matic tape has been loaded, the entire cutting process can be controlled from a central VDU and keyboard unit. After being read from the U-Matic digital

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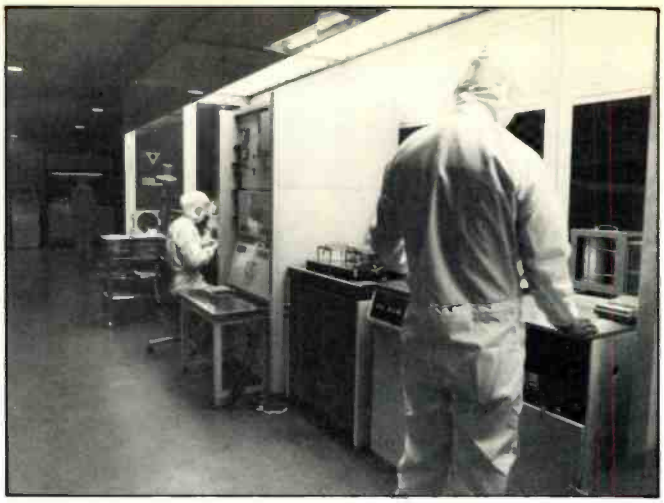
A Sony DMC-1100 Disc Master Code Cutter; Terre Haute will be equipped with two such laser cutting systems.



Glass Master preparation and photo-resist coating in the A-Room at Shizouka.

master, the PQ Code date, in conjunction with a SMPTE timecode track, provides information to the system regarding track numbering and duration. An intensity-modulated laser beam then "cuts" the pattern of pits into a photo-resist layer previously coated onto the surface of the glass master.

The required accuracy of the laser-cutting process can only be described as extremely exacting. For example, each pit etched into the photo-resist layer is 0.5 micrometers (0.02 mil) wide, between 0.8 and 3.5 micrometers (0.03 to 0.14 mil) long, and radially spaced 1.6 micrometers (0.06 mil) apart. Also, the spiral of pits is cut from the inside of the glass master to the outside at a constant linear velocity — in other words, the rotation speed master is varied between 500 and 200 RPM during the cutting process to compensate for the increasing diameter from inside to outside edge.

Following development and etching of the photo-resist material, the glass plate is stored under an inert nitrogen atmosphere to prevent oxidation and additional contamination until it is ready for transfer to the B-Room, and the metal-plating process. (Incidentally, I understand that after careful cleaning and polishing to ensure a true optical flatness, and recoating, the glass plates can be re-used in the laser-mastering process between 10 and 20 times.)

B-Room — Metal Master and Stamper Production

In this area, which in principle bears a close relationship to the corresponding stages involved in the production of a conventional vinyl pressing, the previously prepared glass master (representing a positive form) is coated with a nickel or silver spray to make it electricaly conductive, and then placed in a plating bath. The resultant layer of nickel grown on the glass plate forms the negative metal master from which, by means of a second plating process, a metal mother is produced. And from this second piece of metalwork can be produced the final negative-form metal stamper used in the injection-moulding process.

Plant staff at CBS/Sony's Shizouka facility tell me that each laser-etched glass plate and negative metal master can be used to produce a maximum of five mothers, from which in turn can be grown a maximum of five stampers. So far, practical experience has shown that approximately 8,000 CDs can be taken from each stamper, although Shizouka management concede that it might be possible to pull more disks from each part, should demand be higher during a production run. (It doesn't take much of a mathematician to calculate that, in theory at least, approximately 200,000 CDs could be pulled from one glass master and its resultant metalwork parts. However, since market demand so far has not reached these levels, there has been no reason to test such production potential.)

C-Room — Injection Moulding

In the C-Room, poly-methylmethacrylate (PMMA) granules are loaded into injection machines containing the appropritate nickel stamper, and a clear disk substrate is produced. A total of eight injection machines are available at the Shizouka plant, cycle and production time per CD being on average approximately 15 seconds.

The Compact Discs are then loaded in batches of 100 at a time into a vacuum-evaporator unit, where both surfaces receive a very thin layer of metallic material that serves as a reflective coating for the CD player's laser optical

After development of the photo-resist layer, each glass master is used to grow a negative metal master, and subsequent mothers and stampers in the B-Room (left). Quality-control inspection occurs at various stages in the CD process, including the C-Room (right).

The unique Calrec Soundfield Microphone MK IV

THE FEATURES

- ☐ **Totally steerable both vertically and horizontally in POST PRODUCTION, off tape.**
- ☐ **Variable stereo capsule angle and polar pattern in POST PRODUCTION, off tape.**
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- ☐ **Very low noise performance for the digital era.**
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- ☐ **Level frequency response both on-and off-axis.**
- ☐ **The most accurate polar patterns in the world.**
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THE FACTS

The spherical three-dimensional pick-up of the Soundfield Microphone is such that the phase errors introduced by the capsule spacing in normal microphones are effectively eliminated and the resulting stereo output of the control unit has virtually perfect image placement at all frequencies. The differing frequency responses of the pressure and gradient components of the signal are also corrected, thus giving an equally flat response to both on- and off-axis sounds. These two facts make it possible, for the first time ever, not only to generate exactly signals envisaged by A. D. Blumlein when he first proposed the M/S system, but to extend them into three dimensions.

This spherical representation of the original soundfield allows a stereo signal to be extracted pointing in any direction and of any first order polar diagram. The angle between the two microphones may be varied between 0° (mono) and 180° and the apparent proximity to the original sound sources may also be adjusted.

The control unit also provides a four-channel output signal, known as "B format," which exactly represents the first order characteristics of the soundfield. Recordings stored in "B format" allow the POST SESSION use of all the aforementioned controls. The advantage of being able to set such critical parameters as image width, direction of point and tilt, polar patterns and distance – all in the peace and quiet of the dubbing studio – cannot be over-emphasised. "B format" is also the professional signal format for Ambisonic surround sound and may be encoded directly to domestic transmission and consumption formats.

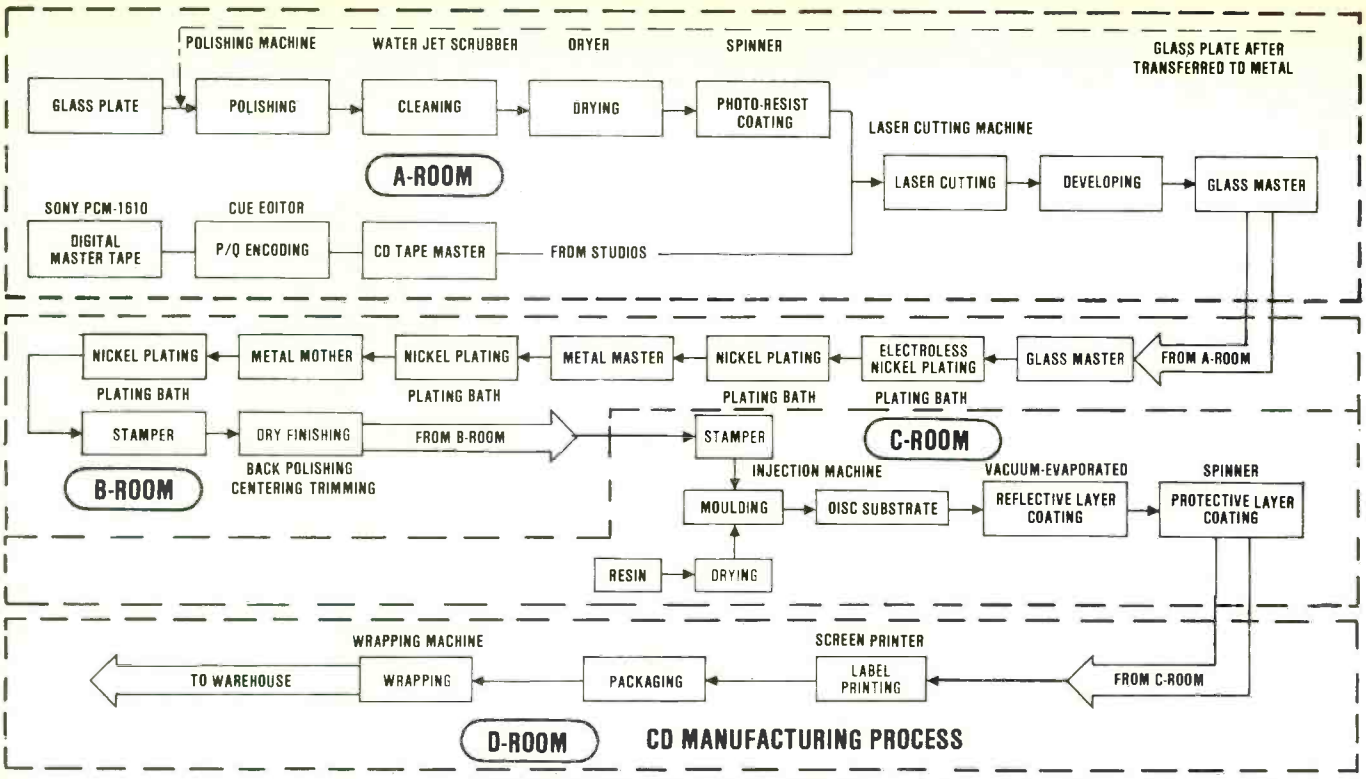
For further information, please contact your local dealer, national distributor or:-

CALREC

Calrec Audio Ltd., Hangingroyd Lane, Hebden Bridge, West Yorkshire, England. Telephone: 0422 842159. Telex: 51311 Relays G.

U.S.A. & CANADA
Audio and Design, Inc., P.O. Box 786, Bremerton, WA 98310
Telephone: 206275 5009 Telex: 152426 ADR USA.

For additional information circle #57



pickup system. Cycle time in the evaporator unit is 15 minutes. From here the CDs are placed on a continuous-feed unit that coats the lower surface with a protective layer, spins each Compact Disc to ensure complete coverage, and then utilizes an ultra-violet lamp to stabilize the coating.

D-Room — Label Printing and Packaging

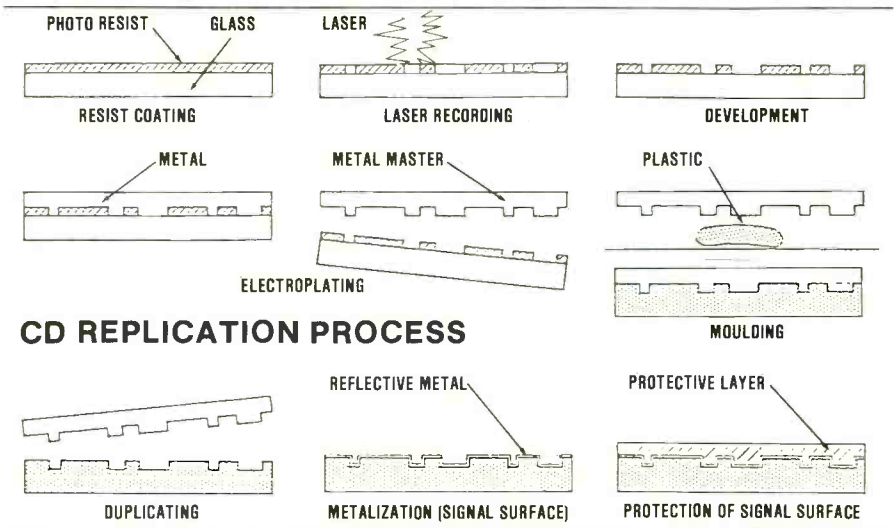
Using specially developed, wear-resistant inks, labelling information is printed directly onto the top surface of each CD. (Remember that the playing surface is on the bottom of the Compact Disc, where changes in the reflective nature of the spiral-cut pits are detected by the CD player's servo-controlled laser optical pickup system.) It is in the D-Room that liner notes, the plastic CD holder, wrapping and packaging material come together, and the final consumer product emerges.

Visual and Automated Inspection Process

With such a high-tech product being produced, the slightest defect in the CD manufacturing process has to be caught as soon as possible, to avoid excessive rejects. The inspection process begins in the B-Room, where the stampers are visually checked for dust; automated equipment also is available to the Shizouka production staff to speed up the inspection process. In addition, an optical system enables the glass master and stampers to be checked for possible high-frequency signal loss during the original laser cutting stages.

Sources of possible defects in the C-Room include foreign materials contaminating the PMMA granules, inconsistencies in moulding, or pinhole anomolies in the vacuum-deposited reflective layer. During later production stages the finished Compact Discs are checked electronically on a microprocessor-controlled CD player/analyzer for frame-error rates, tracking noise, and interpolation errors.

I understand that approximately 2.5% of the total CD production at the Shizouka plant is either visually or electronically checked for potential mastering and/or manufacturing defects. An emphasis will be placed at the new Terre Haute plant, CBS/Sony staff point out, on further automating the inspection process. ■■■



CURRENT DEVELOPMENTS IN DIGITAL STORAGE MEDIA

Although the digital recording era has brought with it many operation and sonic advantages over conventional analog technology, in terms of its application as a storage medium there are still several drawbacks to using magnetic tape. It cannot be denied that tape offers incredible packing densities: one track of 48 kHz/16-bit digital audio data represents, with error correction overheads, a bit stream of around one megabit per second, or 33 kilobits per inch at 30 IPS, which makes it ideal for long-term

archive storage. In terms of quickly finding any point in the data stream, however, magnetic-tape transports are simply too slow and cumbersome to provide anything approaching fast random access to the recorded audio material. The use of data storage technology borrowed from the computer industry —including Gotham Audio's hard-disk-based Systex 300, and Compusonics' new floppy-disk-based mixing/ recording system (described elsewhere in this issue) — can provide one solution, although the costs involved sometimes can be prohibitive.

It is hardly surprising, therefore, that several companies around the world are currently developing digital mass-storage devices that provide data capacity and random-access times comparable to or better than conventional hard-disk systems, but which are less complex, and should sell for a fraction of the cost. And let's not forget that the consumer and professional audio industries represent but a fraction of the potential market for such hardware. Imagine being able to mass produce the electronic equivalent of a text book, dictionary or encyclopedia on an 8- or 12-inch disk, complete with full-color graphics, and which can be random accessed by a computer-based text-management system.

Of particular relevance to the pro-audio marketplace are recent developments in erasable Videodisk and Digital Audio Disk technologies. Despite the fact that the driving force behind these innovations from consumer-electronic giants such as Matsushita/Panasonic, Philips, Sony, Sharp, and others is the inevitable home-user market — allowing high-quality "time-shift" video and digital audio recording, for example — there are numerous applications for the recording studio, film and audio/visual industries. Consider, for example, the creative possibilities offered by the ability to track sound effects onto a cheap, re-usable storage medium, and then play them into a composite mix against SMPTE timecode locations. Or maybe the ability to pre-assemble various sound elements against a master timecode sequence, and then slip each of the independently as the need arises, or to follow picture and/or audio edit changes during the production of a film or video release.

Contemporary areas of developments in high-volume/low-cost data storage include hardware derived from Compact Disc, LaserDisk, and magneto-optical technologies. Described below are some of the more innovative example of current or soon-to-be-released equipment for recording and random-access replay of digital-quality sound.

COMPACT DISC RECORDING TECHNOLOGY

Given the growing penetration of CD players into the consumer marketplace, and the format's extended playing time, 90+ dB dynamic range, and simplicity of operation, Compact Disc record/replay units cannot be far away on the digital horizon. During a recent visit to Sony's research and development laboratories in Japan, this writer had the opportunity to discover first-hand what kind of new digital hardware the company plans to unveil in the not-too-distant future.

In keeping with Philips, Toshiba,

LASER VIDEO, INC. EQUIPPED TO HANDLE SMALL-SCALE COMPACT DISC PRODUCTION USING PHOTOLITHOGRAPHIC PROCESS

As if only to demonstrate that the large European and Japanese record labels may not have the CD manufacturing market all to themselves, a small company based in Anaheim, California, has been undertaking small-scale Compact Disc pressing for several months now, and plans to have a plant capable of mass producing CDs on stream later this Fall. According to the company's president, Wan Seegmiller, Laser Video, Inc. has been working in optical videodisc technology for the last four years, and has developed a photolithographic replication process for CD manufacture. Using this new process, which is said to be quicker and simpler in concept than conventional Compact Disc pressing techniques, the company has produced sample quantities of CDs for several record companies around the country.

Source material for the CD replication process can be either a normal PCM-encoded, U-Matic digital master with accompanying PQ Subcode information of track timings and durations, or an analog master tape. In a process virtually identical to that utilized at other CD pressing plants (described in detail in the main article above), Laser Video first prepares a glass plate coated with a photo-sensitive layer. Then, by laser etching this layer, a positive will be produced. However, instead of producing several stages of metalwork from the glass master, and then injection moulding the CDs, Laser Video simply contact prints the pattern of pits, using a similar UV-light photo-resist and etching process, onto a disk of plastic substrate material coated with a thin layer of metal. (In essence, the process differs little from that already being used widely by the semiconductor industry to manufacture multilayer IC wafers.) The resultant metal-coated disk can be played by any CD player.

While Seegmiller concedes that his company doesn't intend to manufacture CDs in vast numbers, he says that once the new Laser Video plant comes fully on-line it will be capable of producing around two million Compact Discs per year. Further details of this new manufacturing process can be obtained from: Laser Video, Inc., 1120 Cosby Way, Anaheim, CA 92806. (714) 630-6700. □□□

Hitachi, Matsushita, and others, Sony is developing CD-DRAW (Digital Read After Write) technology to enable an analogy signal to be digitally recorded with a laser beam onto a spinning plastic disk similar in size to the present Compact Disc format. Sony's effort is centered around using localized temperature increases to cause a permanent phase transition in a hybrid alloy, from an amorphous to a crystalline state. Without delving too deeply into the somewhat complex material-science processes involved, in essence the new generation of CD-DRAW decks will utilize a high-power laser to heat local areas of a thin antimony-selenide metallic film or recording layer coated on both sides of a plastic disk spinning at 900 RPM, "to produce two binary alloys that melt into one phase of a four-element alloy," according to company scientists. The resultant phase change causes a three-fold increase in the reflectivity of the laser-exposed region, which then can be read by a CD optical pickup.

During the one-time recording process, the recording layer and a second heat-absorbent layer of a bismuth-telluride metallic film are evaporated onto the poly-methylmethacrylate (PMMA) substrate material. (The heat-absorbent layer also intensifies the optical change in the recording layer, and helps reduce the required laser power to around seven milliwatts.) Since a relatively high recording temperature of 120°C is used, the CD-DRAW is said to be unaffected by thermal changes encountered in normal storage conditions. And because the phase transition occurs over a narrow 20-degree temperature range, the edges of the pits produced in the recording layer are said to be sharp,and the length rigidly determined — enabling the technique, Sony claims, to be used not only for digital audio recording, but also analog/FM recording, similar in concept to present-generation optical videodisks.

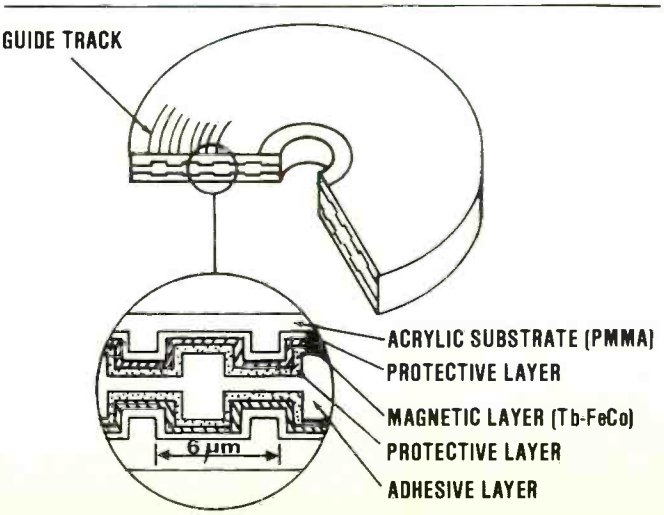
At present, Sony is working on prototype record/replay decks that will utilize two diameters of the double-sided CD-DRAW: a 20cm (7.9-inch) version capable of holding six gigabits of digital information per side (approximately one hour of 44.1 kHz/16-bit stereo audio); and a 30cm (11.8-inch) version that will hold 15 gigabits. I understand that the new generation of decks will be able to replay "conventional" CDs, as well as read/write to CD-DRAW.

Apart from its obvious applications for digital audio record/replay, CD-DRAW and its future variants also look set to replace floppy- and hard-disk systems for mass-production and storage of computer data, software, and similar utilities. (Assuming that "blank" CD-DRAW disks are priced low enough, disposable data bases, such as dictionaries, parts inventories, and mailing lists, might be feasible, not to mention mass production of program software.) However, two further developments of Compact Disc technology are configured specifically for software and data storage applications. Connected to a suitable control computer, a simplified Compact Disc running the CD-ROM format recently proposed by Philips and Sony will provide access to over 550 megabytes of 8-bit data from a one-hour disk, and should offer enormous potential for such fields as data-base storage, electronic publishing, and audio-video production.

There is insufficient space in this article to detail every possible aspect of the new CD-ROM format, although a couple of example should provide a good idea of its capabilities: in graphics mode, a single disk could hold 18,000, 320- by 200-point pictures; in still-frame mode, 3,600, 420- by 240-pixel frames; in alphanumeric mode, 112,000, 80-column by 60-line pages; and, of particular interest here, one hour of four-channel, 8-bit "Adaptive PCM" digital audio recording at a sampling frequency of 33.075 kHz. Access time for prototype CD-ROM players is quoted to be around three seconds to any point on the disk. In addition, Sony points out that, with a suitable hardware controller, any combination of modes can be intermixed. This would enable, for example, an hour

The new Panasonic TQ-2023 Optical Memory Disk Recorder.

Several companies are currently developing erasable magneto-optical disk transports, capable of holding up to 12 gigabits of data. Shown right is a section of the Sony/KDD disk format, which uses a Te-Fe-Co magnetic film alloy as the recording medium.



of two-channel digital audio to be recorded simultaneously with graphics, still-frame, or program data. (Obviously, the intermix format will halve the total capacity of each respective mode.)

And finally, several manufacturers are perfecting methods of utilizing the "spare" user bits currently not accounted for in the CD subcode areas. Up to 3% of a Compact Disc's total data capacity — amounting to the equivalent of around 20 megabytes of data — is still free for recording up to 30,000 computer graphics and still-frame pictures, which might comprise musical notes and/or lyric sheets for the accompanying digital audio.

PANASONIC TQ-2023 OPTICAL DISK RECORDER

Heralded by its manufacturer as the world's first commercially available optical memory disk recorder (or OMDR, to use the jargon) that can record and playback full-motion or frame-by-frame color video signals with accompanying FM-encoded two-channel audio, the TQ-2023 will record up to 24,000 video frames in still function, and up to 13.3 minutes of real-time signals. (This latter capacity is available in a "spiral-disk" mode; a "concentric" disk can store 15,000 video frames, or eight minutes of continuous picture and sound.)

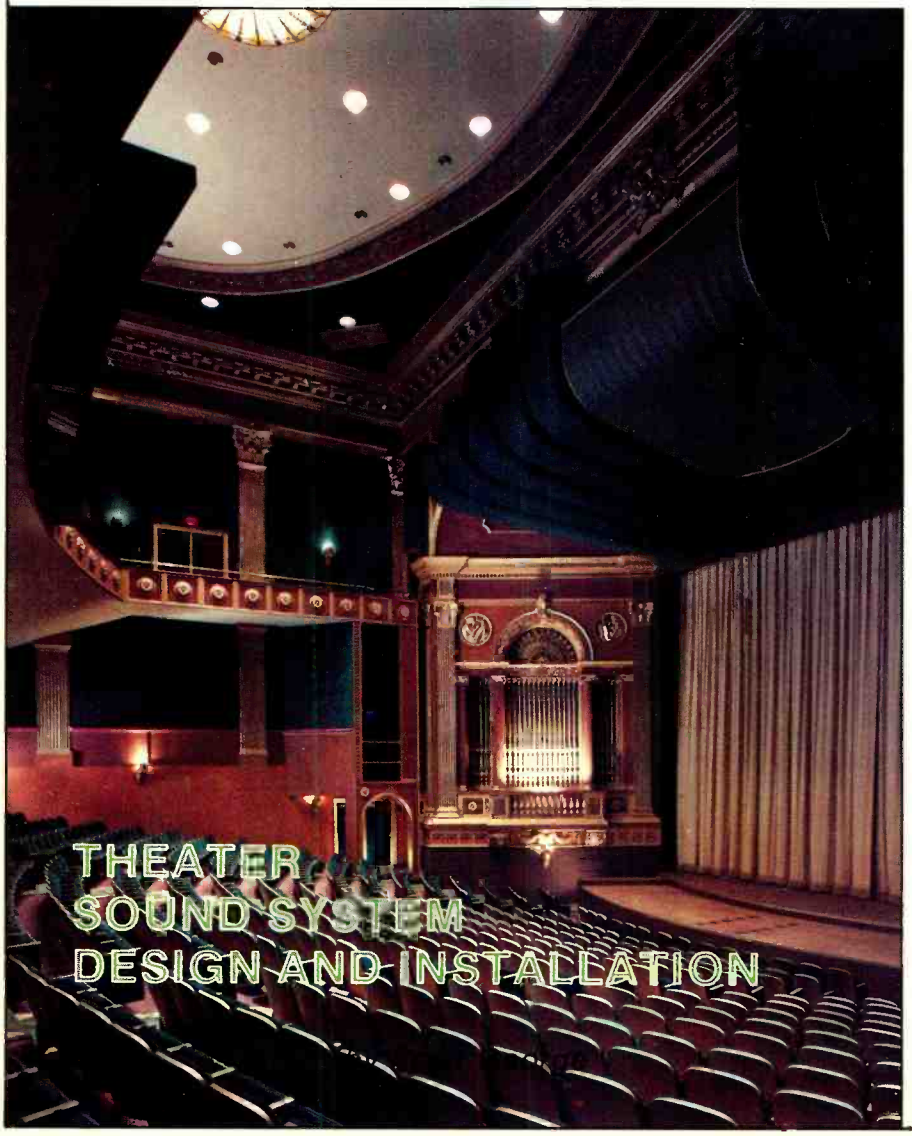
Utilizing semiconductor laser technology to one-time record and playback from an eight-inch optical disk, fastest access time to any video frame via front-panel, remote control, or built-in RS-232C computer interface is a quoted 500 milliseconds. Panasonic says it should have an erasable version capable of re-recording video and audio signals on the market some time next year.

Obviously such an audio/video recorder has enormous potential in the editing, industrial, and corporate video fields, but what of possible pro-audio applications? Several studios and production houses are currently experimenting with computer-controlled LaserDisc players, to enable individual pre-recorded FM-encoded sound elements to be cued and played back from a SMPTE timecode location. Despite the fact that 3M, for example, can provide custom-mastered LaserDiscs at a reasonable cost (around $300 being a typical price tag for one-off pressings), a turn-around time of even a few days can cause production delays. Devices like the new TQ-2023 OMDR will enable sound effects and similar material to be recorded when and where they are required — possibly from a central tape- or hard-disk-based library of digital effects and music tracks that can be downloaded to local, temporary storage before the start of a sweetening or mixing session. And while it could be argued that the FM-encoded soundtrack of a LaserDisc cannot match the quality of digitally recorded audio, its 75+ dB signal-to-noise ratio, frequency response and distortion performance should be more than adequate for all but the most demanding sessions.

MAGNETO-OPTICAL ERASABLE DISK TECHNOLOGY

Several companies, including Sharp, Matsushita, NHK, Philips, 3M, Sony, and Xerox are currently developing large-capacity erasable magneto-optical disks that allow data, including digital audio, to be recorded and then erased for subsequent re-use. One format, co-developed by Kokusai Denshin Denwa (KDD) and Sony, and being demonstrated in prototype form during my recent Japanese visit, is capable of holding up to 12 gigabits of information — equivalent to between 2,000 and 3,000 NTSC video pictures — on a double-sided, 8-inch disk running at a constant angular velocity of 1,800 RPM. The recording layer comprises a thin amorphous magnetic film of terbium, iron, and cobalt alloy, the magnetization of which initially is vertically aligned. Explained very simply, a small "bias" coil located in the center of a recording laser beam produces a magnetic field when energized that enables the laser to alter the polarity of a selected signal pit etched in one of 28,000 tracks running across the disk's surface. The data can then be read by a laser pickup assembly similar in design to that utilized in conventional Compact Disc and CD-DRAW transports. ■■■

AUDIO FOR THEATRICAL PERFORMANCE



There are usually three types of sound systems to be found in today's theatres. Two of them are likely to be built-in: What I would refer to as Amplification and Sound Effects Systems. The third — the Show Sound System — is likely to be tailored to the specific electro-acoustic demands of the production, and may well vary from one show to the next. In most recently built, or rebuilt theatres, the two permanent systems probably would have been designed by an acoustical consultant or sound systems engineer. These systems will be carefully selected to suit the room acoustics of the theatre, regardless of the type of production being presented. Although the sophistication of the built-in systems will be controlled primarily by the theatre-equipment budget, it must be broad enough in scope to accommodate the needs of all the theatre's potential users.

Ascendance of the Sound Designer

The Show Sound system, however, is likely to have been installed by the burgeoning group of production technicians knows as "Sound Designers." Once upon a time, the sound designer was an occassional member of the stage production design team that normally includes the set, lighting, and costume designers for, say, a large scale musical. In the last 10 years, however, more types of stage productions have made use of the services of a sound designer. In general, the sound designer is more likely to set up temporary sound systems that meet the specific requirements of the production, rather than the acoustical needs of the house itself. The selection of equipment and the location of loudspeakers are likely to be very subjective decisions that will vary considerably from one designer to the next.

This article is intended to provide an assessment of the state-of-the-art regarding theatrical electro-acoustics, and will include the roles of the various designers along with a summary of what types of systems are being put to use. The majority of the theatres being built today are still set up in the stage-proscenium-audience configuration, and so the following commentary will apply to this type of space (rather than, say, a thrust stage or theatre-in-the-round configuration).

Loudspeaker Positioning

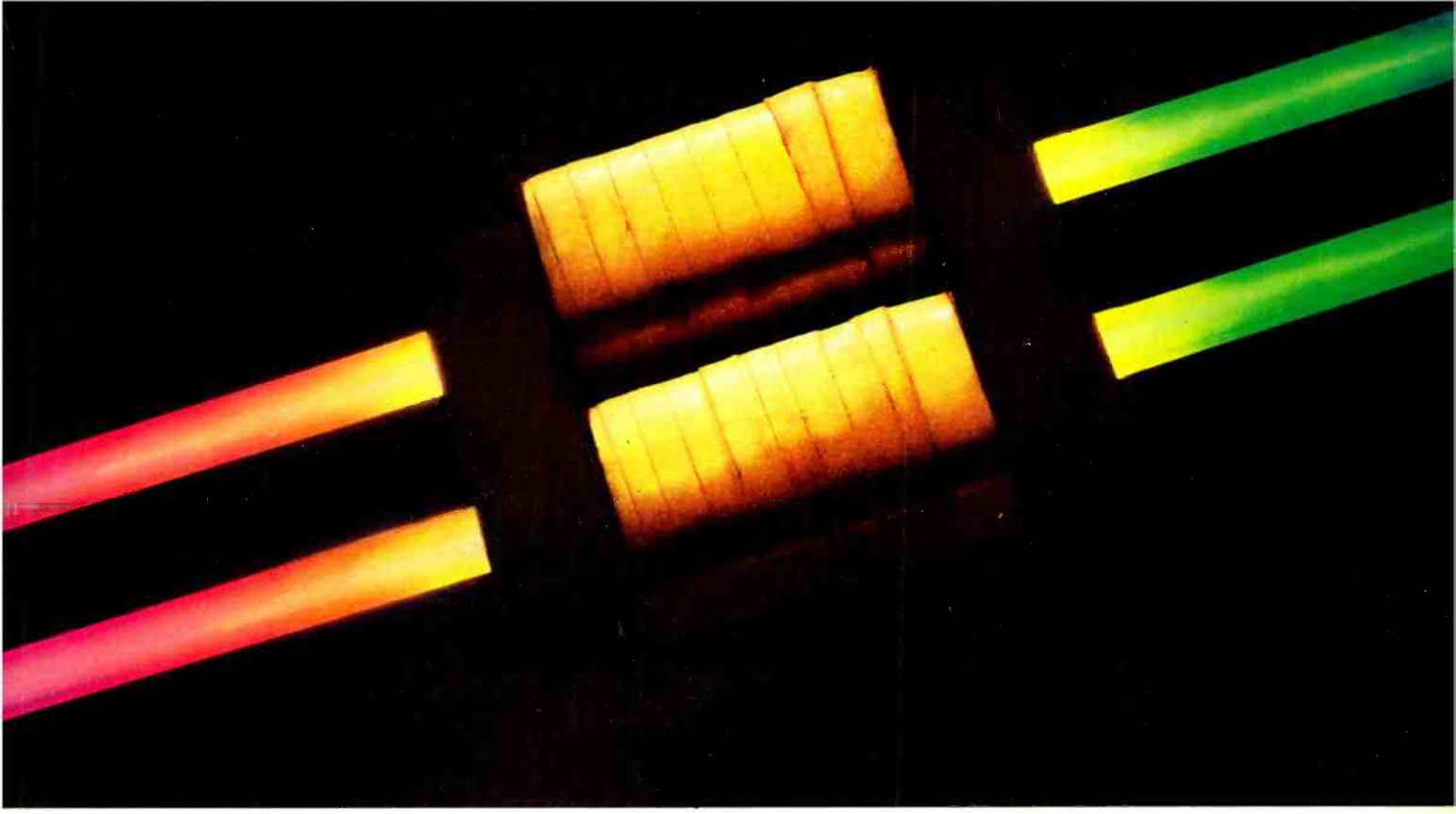
The loudspeaker cluster, located centrally above the proscenium, is the preferred position for house-sound reinforcement, since the hearing mechanism is less able to locate vertical separation of sound sources than it is those which are horizontally separated (as happens when loudspeakers are located on either side of the proscenium). As a result, sound from the overhead cluster is more likely to appear to have originated at the performer on stage. Some engineers like to spread out their loudspeakers across the proscenium ("Broad Front") for a variety of special reasons, but use of a central overhead cluster is the favorite. Occasionally, when the budget is right, it is possible to mount the cluster on a retracting platform that disappears flush with the ceiling when not in use. At the central overhead position, the cluster can be exposed for a "High-Tech" look, but is more usually concealed behind an acoustically transparent fabric or grille covering, held in a cage within the ceiling.

Careful juggling is needed between the three members of the design team: the *sound system engineer,* who requires no restriction of the required dispersion patterns; the *acoustician,* who wishes that the transparent area could be minimized so that remaining "hard" surfaces could be maximized to send (carefully angled and shaped, of course) early reflected, unamplified sound energy into the audience (and under that balcony); and the *architect* who detests the thought of *any* kind of patch in his ceiling, especially *there* where everyone is looking. (In truth, of course, we all hope that, for the best part of the time in the theatre, the audience will be looking *through* the proscenium, not *above* it — a concept that is, sad to say, lost in most architects . . .)

While such juggling may seem problematic, it is a simple obstacle compared with facing the engineer involved in the "renovation/rehabilitation/restortion" of a treasured historic theatre. Here, it is necessary, more often essential, to become a magician in order to have house amplification without loudspeakers, or at least "invisible" ones. In the days when theatre building was an art instead of a trade,

— the Author —

Peter George is president of Peter George Associates, a New York-based firm of acoustical and theatre consultants. He has recently completed a study for the Broadway Theatre Owners (Shubert, Nederlander, Jujamcyn) of the acoustics of broadway theatres, as one of the many complex issues being discussed currently regarding landmark designation and development of a Broadway Theatre district. He has designed many sound systems for theatres such as the Peck Pavilion in Milwaukee, the Maguire Theater at Westbury, the Roger Stevens Center in Winston-Salem, Theatre of the Arts in Sarasota, and the Paper Mill Playhouse in Millburn, New Jersey.



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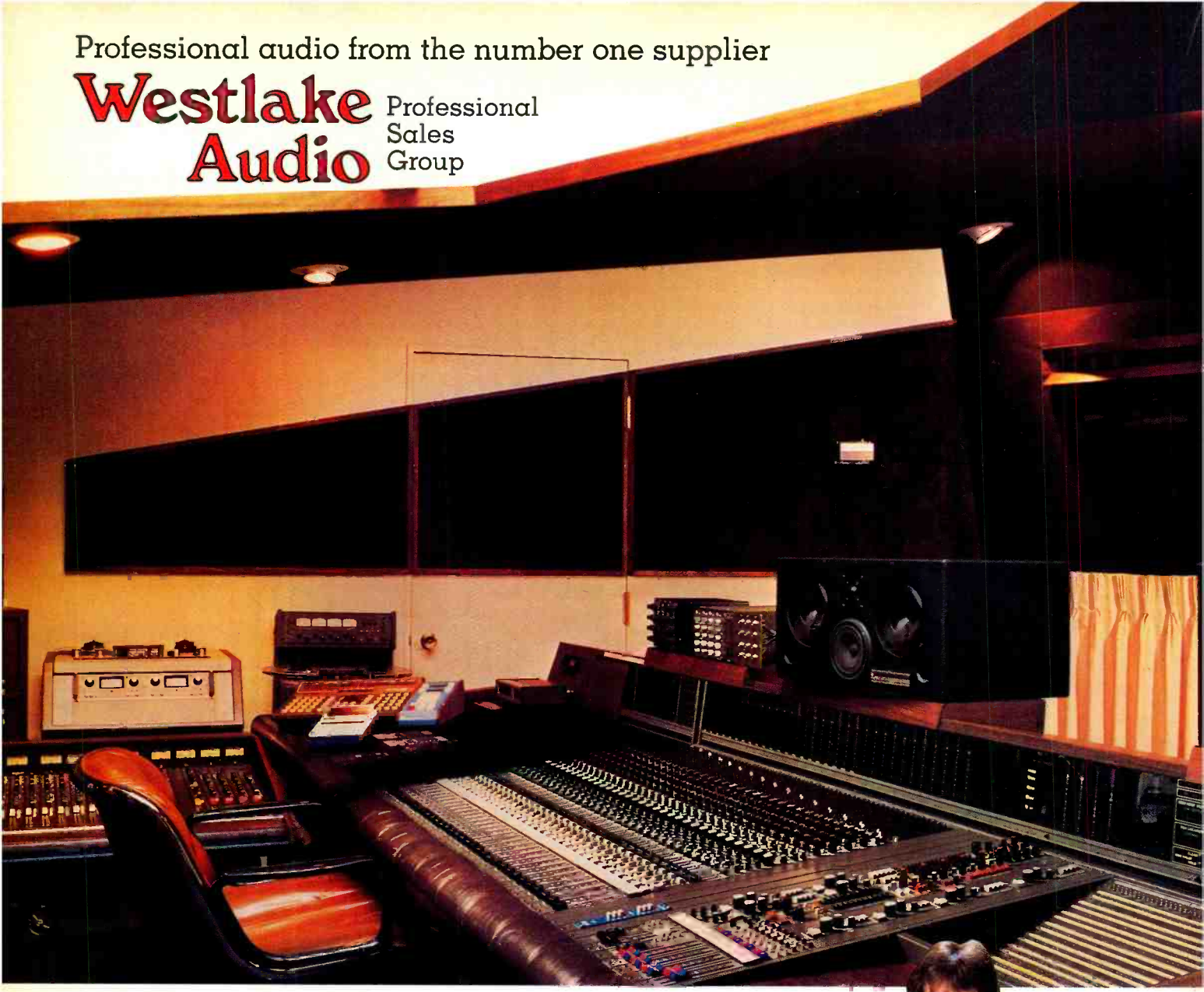

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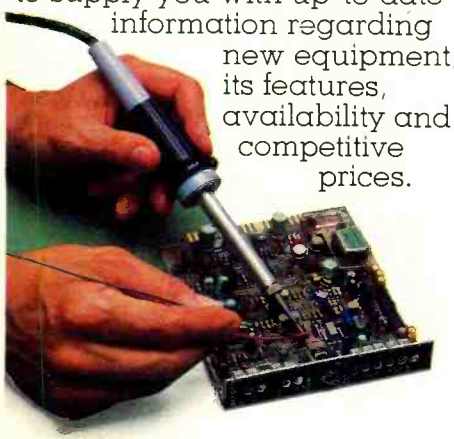

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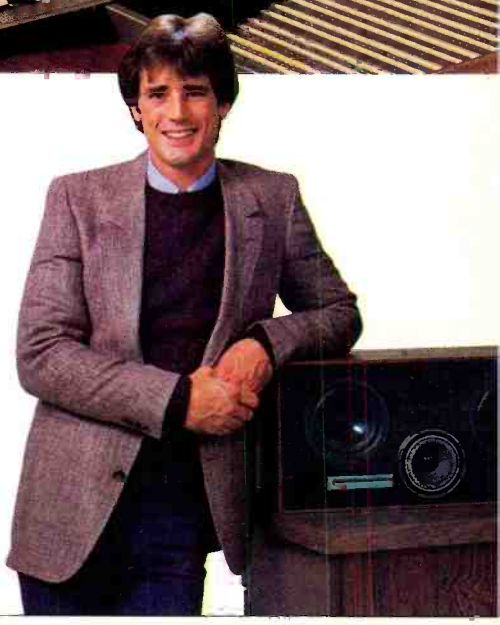

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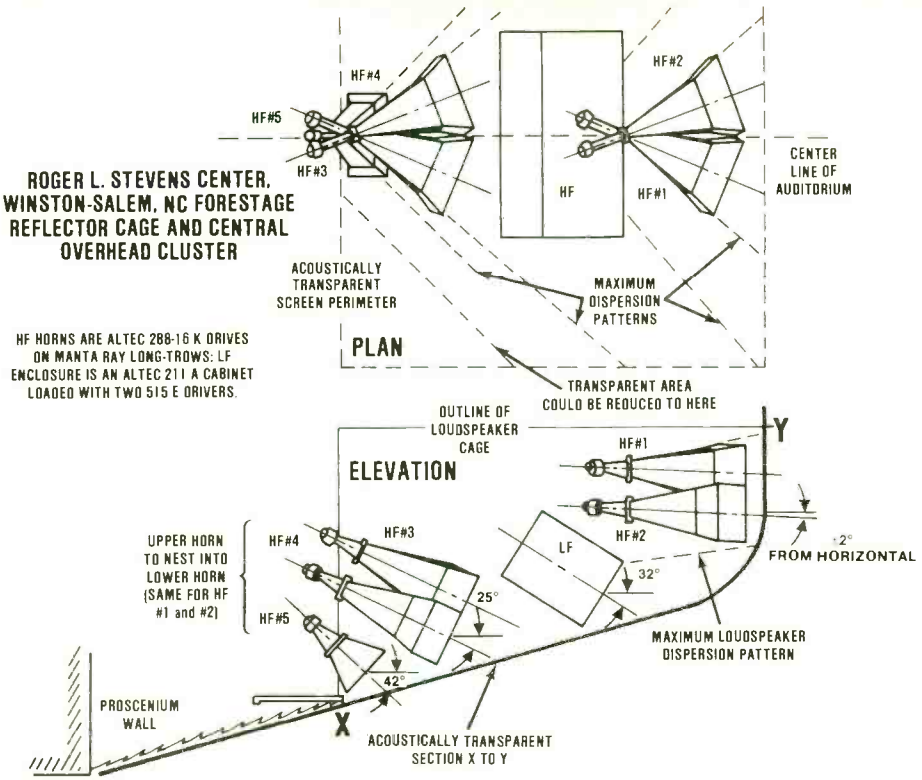
THEATRE SOUND SYSTEM DESIGN



many wonderful devices were created from plaster that were intended to be located above the proscenium in an attempt to distract the audience; I call it the "Bare-breasted-nymph Syndrome." In those days, performers had professional lungs to fill a house (although seat counts of over 1,800 were rare until the advent of the movie palaces). Nowadays, unfortunately, our stages are over-run by television and film actors, latter day members of the "James Dean School for Mumblers," who cannot be heard beyond the third row unless there are footlight cardioids, mike mice or body mikes to comfort them. And, of course, if there are microphones, somewhere there must be loudspeakers (where there's smoke there's fire?)

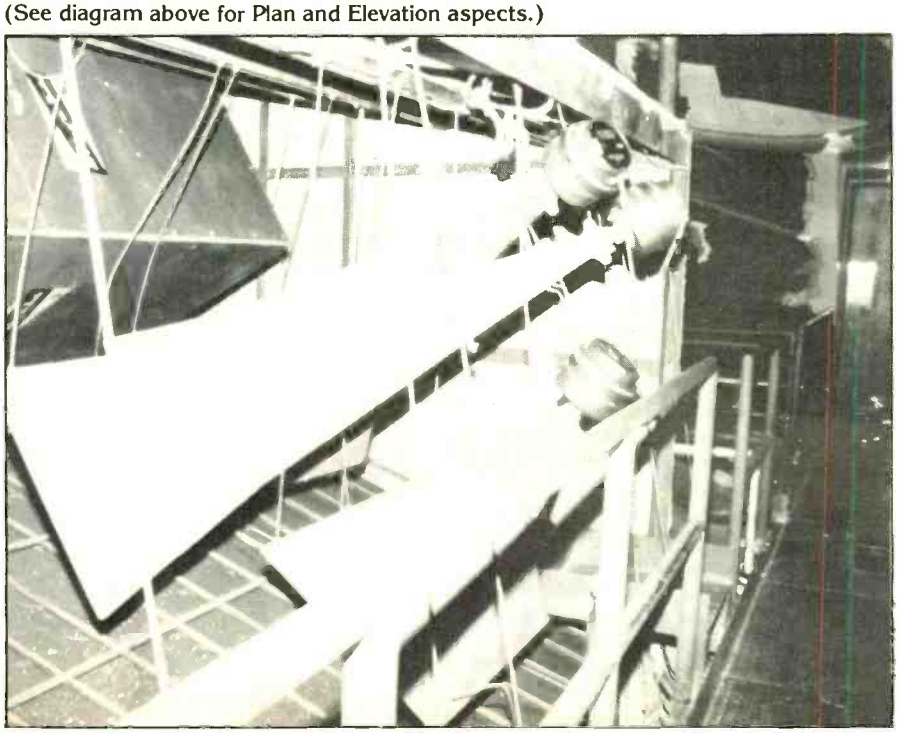
It is somehow, inexplicably, less offensive to the Owners and Guardians of Our Design Heritage to have speakers sitting at stage level or mounted on the proscenium wall, left and right, or peering out of otherwise useless boxes, than to have a "tastefully" designed cluster cage mounted overhead in a central position. Unfortunately, with left and right speakers, a patron sitting off center by more than a few degrees is treated to the show three times over, in rapid succession: first, by the amplified sound from the nearest loudspeaker; then, by a weaker, live signal from the performer on stage, and finally, by a signal from the loudspeaker on the other side of the stage. (Okay, so what if it is 12 dB down in level; it can be heard because it's still louder than what the Mumbler is putting out.)

Dead Spots Under the Balcony

Having solved this one, if he can, the sound system engineer approaches hurdle #2: How to mount the distributed loudspeakers in the balcony soffit without visually betraying their presence. With luck, the original design of this piece of treasured, historic ceiling includes some curlicue grillework through which air is rather crudely dumped into the audience. This being the case — which it rarely is — a midget electrician is employed to crawl into the plenum to locate the conduit and backboxes. By some miracle, the grillework occurs at all the right spots so that 50% overlap of dispersion patterns is possible at the seated ear height of the complete audience squashed together in the narrow end of this wedge formed by the sloping main floor and balcony soffit. Now low-level, delayed sound can be employed that makes the stage performance more audible.

But wait, you say, this combination of architectural circumstances never occurs. What then? Search me — something has to give. There is a partial solution that involves loudspeakers mounted under the balcony edge, facing into that wedge: but more about this below.

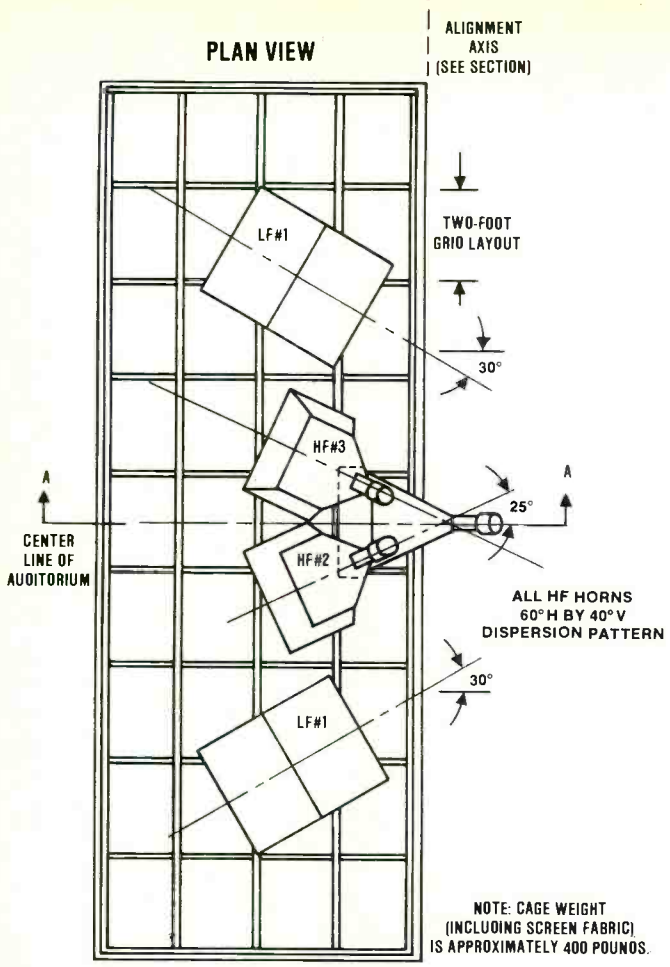
Central overhead cluster of Altec Manta Ray long-throw horns mounted above the forestage cage at the Roger Stevens Center for the Performing Arts, Winston-Salem. (See diagram above for Plan and Elevation aspects.)

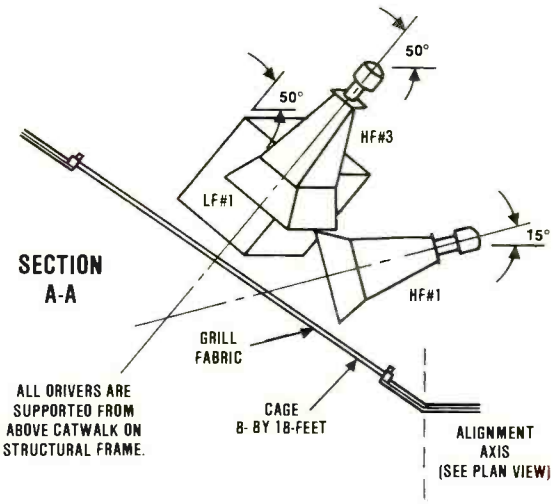


Peter George

To return to the modern theatre, the normal cooperation of acoustician and sound system engineer has obviated the need for under-balcony loudspeakers, since all seats have been designed (it is hoped) to be in full view, at all levels, of the central overhead cluster masterfully concealed in the forestage ceiling. In the last 10 years, the state-of-sound-system-art has been considerably advanced both in the equipment (specifically in the introduction of constant directivity horns from Electro-Voice, Altec, and JBL), and in the science so that clusters and arrays can be designed with computerized precision. (Bar codes printed on the bottom of technical sheets from Altec, for example, points the way to the future in this respect.) Calculations involving house reverberation, critical distance, directivity, equivalent acoustic difference, acoustic gain etc., can be performed using microcomputers with soft ware programs devised by a variety of sources. Altec, the Audio 41 Club (Box 37017, Tucson, AZ 85740) and PPC (2545 West Camden Place, Santa Ana, CA 92704) have programs available for the Hewlett Packard HP-41CV. JBL has developed system design programs for use with IBM PC computer.

The general-purpose theatre or auditorium will also require a special movie amplification system, with its own stage





PAPER MILL PLAYHOUSE MILLBURN, NEW JERSEY CENTRAL OVERHEAD CLUSTER LAYOUT, CAGE, AND INTERNAL DETAIL. (ALL DIMENSIONS AND ANGLES OF LOUDSPEAKER ELEMENTS ARE PRELIMINARY, FINE-TUNING BEING CARRIED OUT ON SITE BY SOUND SYSTEM CONTRACTORS.)

THEATRE SOUND SYSTEM DESIGN

floor loudspeaker position(s), circuitry and amplification originating at the projectors. Theatre (and, more particularly, cinema) owners have suddenly woken up to the need for improved audio with their screen images (and now, here comes Audio-for-Video). There are few major films released these days that are not made available (and most touted thus) with multiple-channel Dolby Stereo soundtracks. (Even Lucas film has entered to the cinema-sound business, although installation of its Tom Holman-designed THX Theatre Sound Reproduction System is probably best suited to permanent cinemas rather than multiple-use theatres; full details are to be found in the December issue of *R-e/p*.)

A typical cinema-style system can provide five behind-the-screen channels, and two surround channels for replaying 35mm optical and 35/70mm optical/magnetic soundtracks. In addition to normal amplification, it is likely to include a two-channel subwoofer power amplifier. The system will interface with Dolby CP-50R or CP-200 and MPU Cinema Processors to ensure correct replay of Dolby Stereo Soundtracks, on the appropriate full-range monitoring curve.

What's in the Control Room?

Attention can be turned to the control and amplification equipment housed in the control room or booth created specially for that purpose. Selection of power amplifiers must first be made to drive the cluster, bi-amplified or tri-amplified as the case may be. The use of incremental power amplifiers in professional systems allow the rapid replacement of expired cards, although preventing, at the same time, as wide a degree of control on each driver card as most engineers would want. Whether space-economic incrementals or stacks of regular amplifiers are used, these units represent only one part of the amplification section. Next, the special needs of the Sound Effects system must be considered, with one amplifier per channel, and sized for the large power requirements per sound effect channel. Basically, the greater the flexibility, the better use this system is to the designer.

It is usual to provide as many loudspeaker receptacles or sockets on stage, on the forestage, within the house and on the catwalks as the budget or the sophistication of the theatre operation will allow. These receptacles are the type that receive twist-lock-type plugs to ensure they remain plugged in for the duration of the show! Since it is usual to run a separate cable to each driver, rather than multiplex cables, this can result in many channels being fed in conduit to the amplifier rack located in the control room. And because sound system design has not yet caught up with the lighting system dimmer-per-circuit concept (cost being the only tangible reason), such channels will probably terminate in a multiple-hole patchbay. The requisite number of single- or dual-channel amplifiers then will be selected to service this network of channels. (To facilitate complex routing and channel assignment, electronic matrix patching units are available from Ramko Research, among other manufacturers.) Rack-mounted signal processing equipment typically will include graphic equalizers, whose settings will be fixed, with face plates bolted (or preferably welded!) in place for the house Amplification System, and patchable for the Sound Effects System. Other equipment to be included depends on budget and sophistication, but would normally include compressors, limiters, noise reduction, delays, reverberation enhancement (see below), etc., from a variety of companies, including Orban, Lexicon, dbx, DeltaLab, etc.

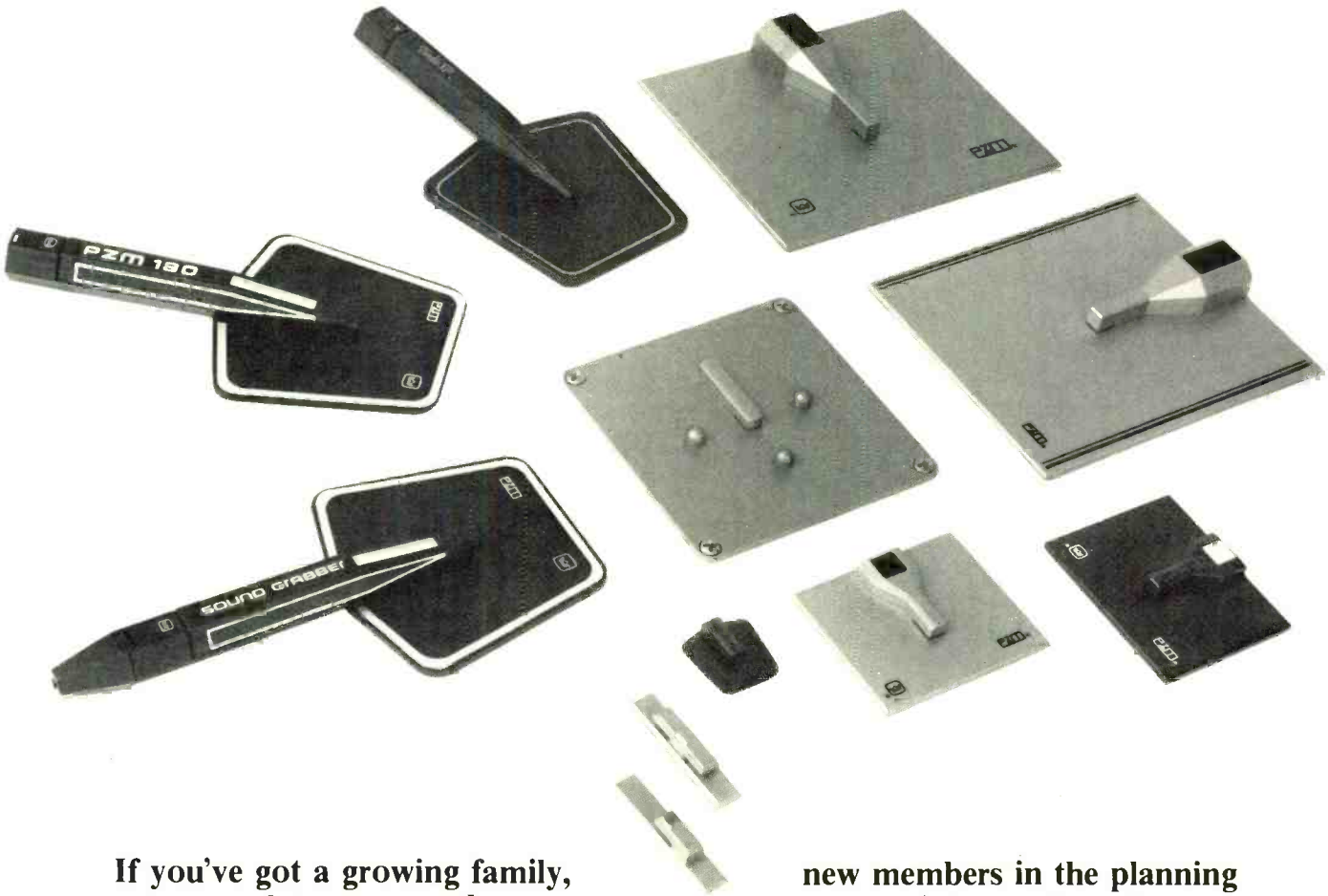
Yamaha PM-512 console in sound booth, Paper Mill Playhouse, Millburn.

Carl Christianson

Program Source Items

Inputs to the system include microphone channels, tape machines, disk players, etc. The number and location of microphone

Family Portrait



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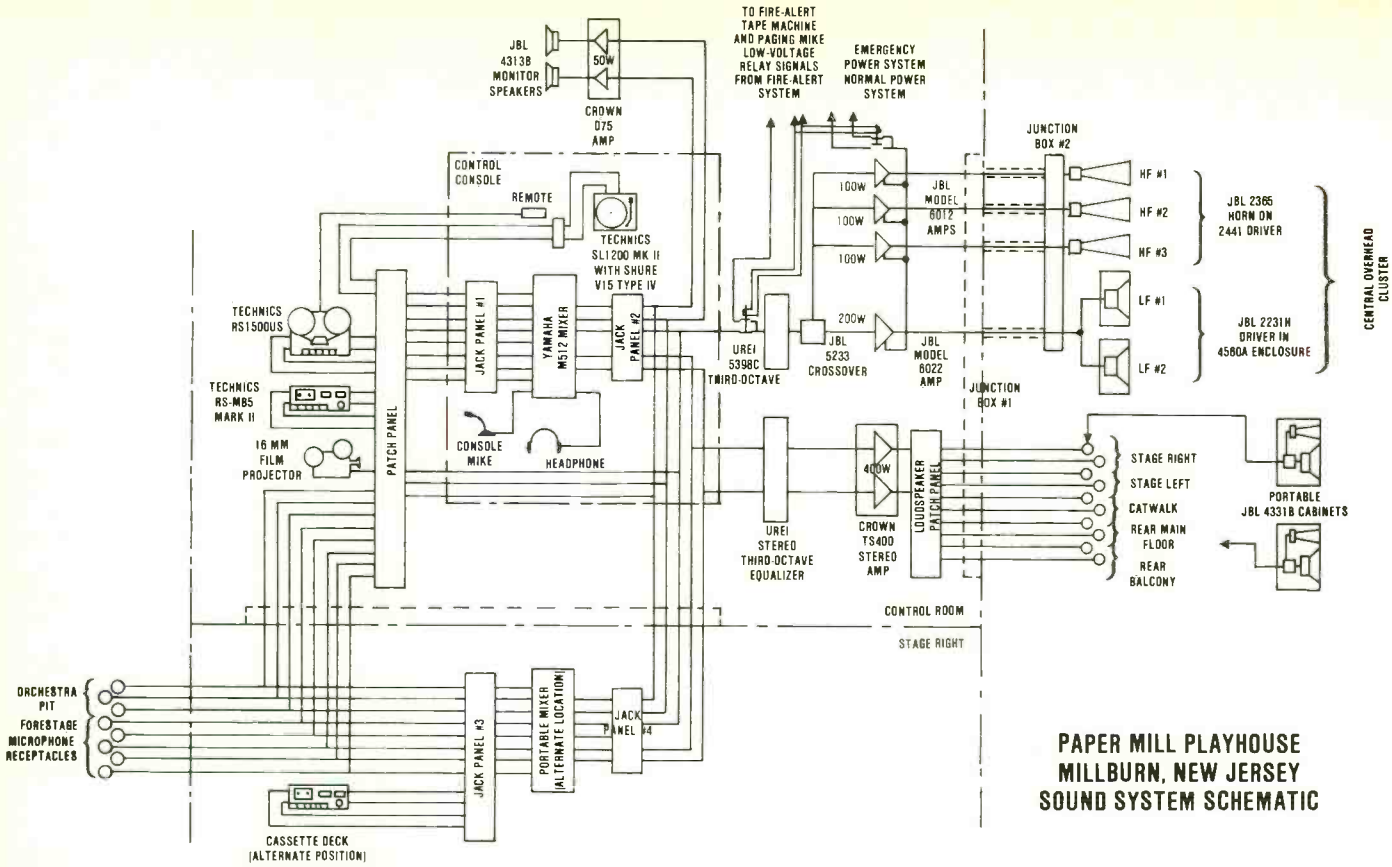


PAPER MILL PLAYHOUSE MILLBURN, NEW JERSEY SOUND SYSTEM SCHEMATIC

THEATRE SOUND SYSTEM DESIGN

receptacles again depends on budget and sophistication. Normally, there will be at least half a dozen receptacles located across the forestage, and an equal number in the downstage edge of the pit lift to take over when *it* becomes the forestage. There should be a good complement of receptacles, located stage left and right and upstage, for "live" effects; the stage manager needs one, as does the Maestro in the pit. Mike snakes (multiway cables) complete the flexibility for multiple-miked shows.

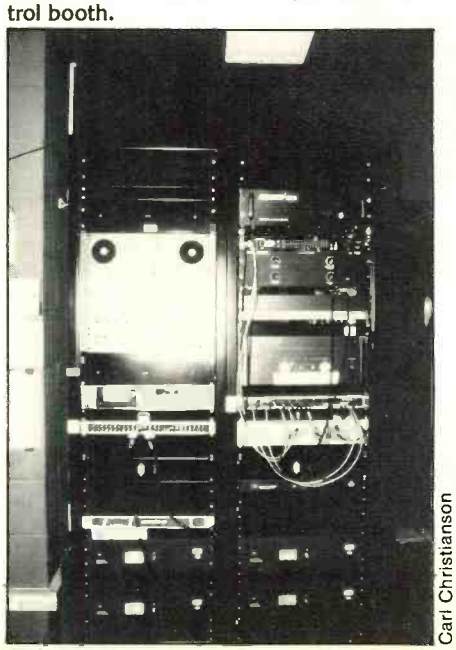
It is preferable to include a receptacle in the house to be used by the director during rehearsals, and/or at the house mix location. If the nature of the likely production calls for much circulation of the actors within or around the house, then receptacles in the walls of vomitories (the tunnel-like entrances into the audience area), or catwalks, etc. are also required. And, of course, the operator in the sound-control booth needs a microphone for announcements. Microphones commonly used during theatrical productions include Shure or E-V dynamics, and Neumann, AKG or Sennheiser condenser models.

As a program source, the reel-to-reel tape machine from such companies as Tascam, Ampex, Otari, ReVox, Technics, and Telex still reigns supreme, although the improved specifications on cassette decks allow interface with professional systems under certain undemanding circumstances. (Such cassette transports should also be capable of record as well as reproduce functions for archival recording purposes.) The Sound Effects system requires a tape machine that will accommodate large diameter reels — 10-inch minimum — to allow easy cueing of pre-recorded material. There is also still place for a turntable, favorite brands appearing to be Technics, Thorens, Lenco or EMT with, say, an SME arm and Stanton cartridge. Digital sound has not yet impacted the sound amplification scene, although it is just a matter of time.

House-Mix Location and Mixing Console

The biggest change in control equipment for theatrical presentation has come from the recording and concert-tour world, in terms of the wholehearted adoption of the multiple-channel mixing console. Not too long ago, a mixer was something that occupied 5½ inches of rack space, and contained a half-dozen knobs. Now it is not uncommon to see 16- 24- or 32-channel boards in the booth, or out at the house-mix location (more about this below). In fact, a house with only a 12-by-2 board is positively deprived. The Yamaha PM1000 or -2000 is most commonly seen, but it is not unusual to find boards from such companies as Soundcraft, Harrison, Audioarts, Neve, etc. For smaller budgets, the modest sized Panasonic RAMSA, Yamaha, Tascam or E-V TAPCO boards are common.

Sound system rack of signal processing equipment in Paper Mill Playhouse's control booth.

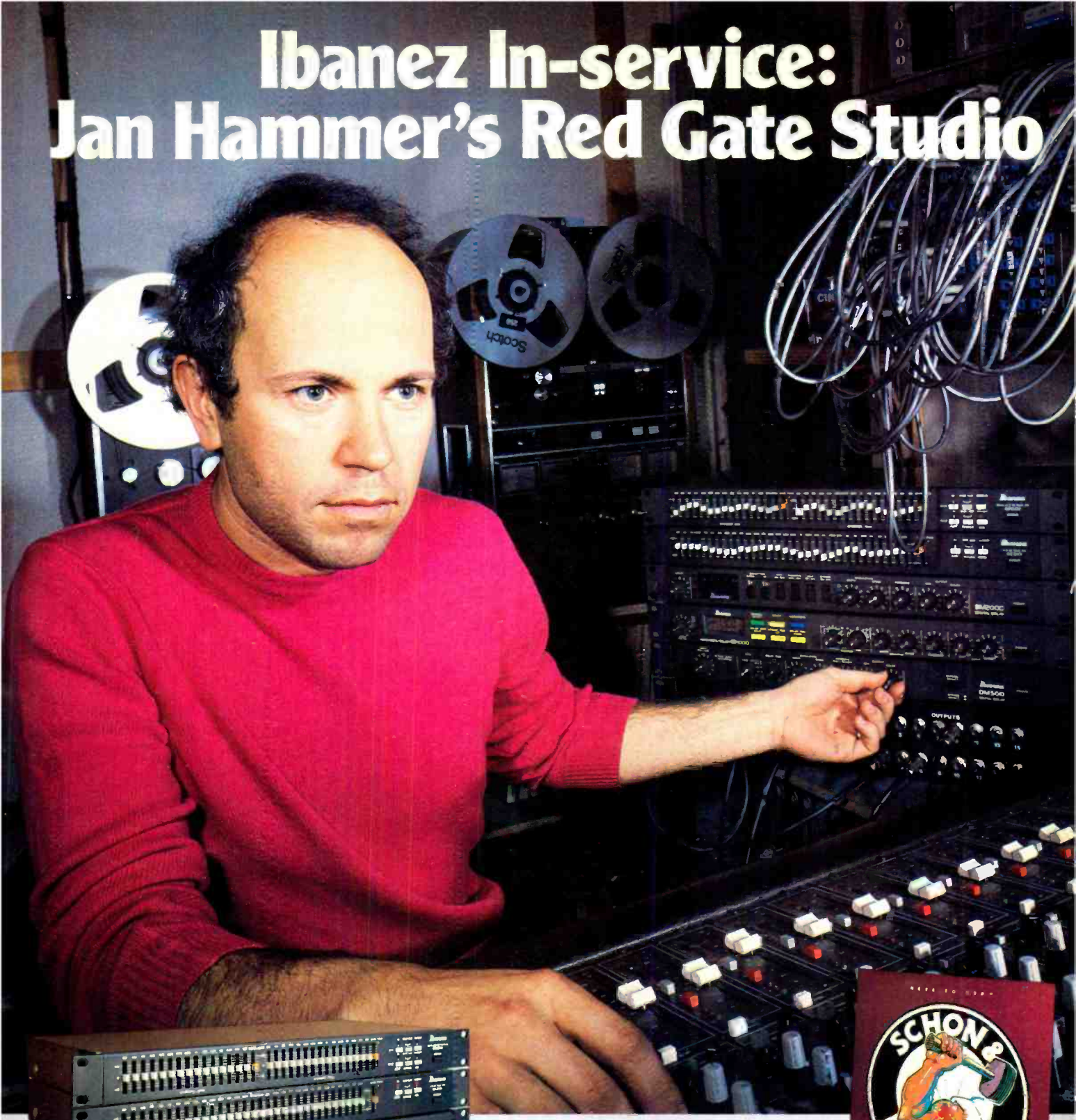


Carl Christianson

The (relative) portability of such multi-channel consoles calls for multiple receptacles so that mixing can be done not only in the control room, but at a house-mix location or on-stage. The mention of house mixing leads me to another subject that is particularly controversial when dealing with a "historical treasure." It is bad enough trying to mix house sound from a control room where the windows open to provide less than four square feet of open area (we usually shoot for at least 16 square feet, and preferably 20 to 24 square feet), but it is even worse when that control room is at the back of 3,000- to 4,000-seat movie house, and possibly not even within the -6 dB dispersion pattern of the horns. Naturally, the operator wants to be out there in, of all things, the area of the best seats and, worse yet (say the Owners), in full view of everyone. Think of all those blinking red lights that will distract from the bare-breasted nymphs!

It is not uncommon in existing Broadway houses to see the board located ignominiously at the rear of the main floor audience area. Occasionally, *very* occa-

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



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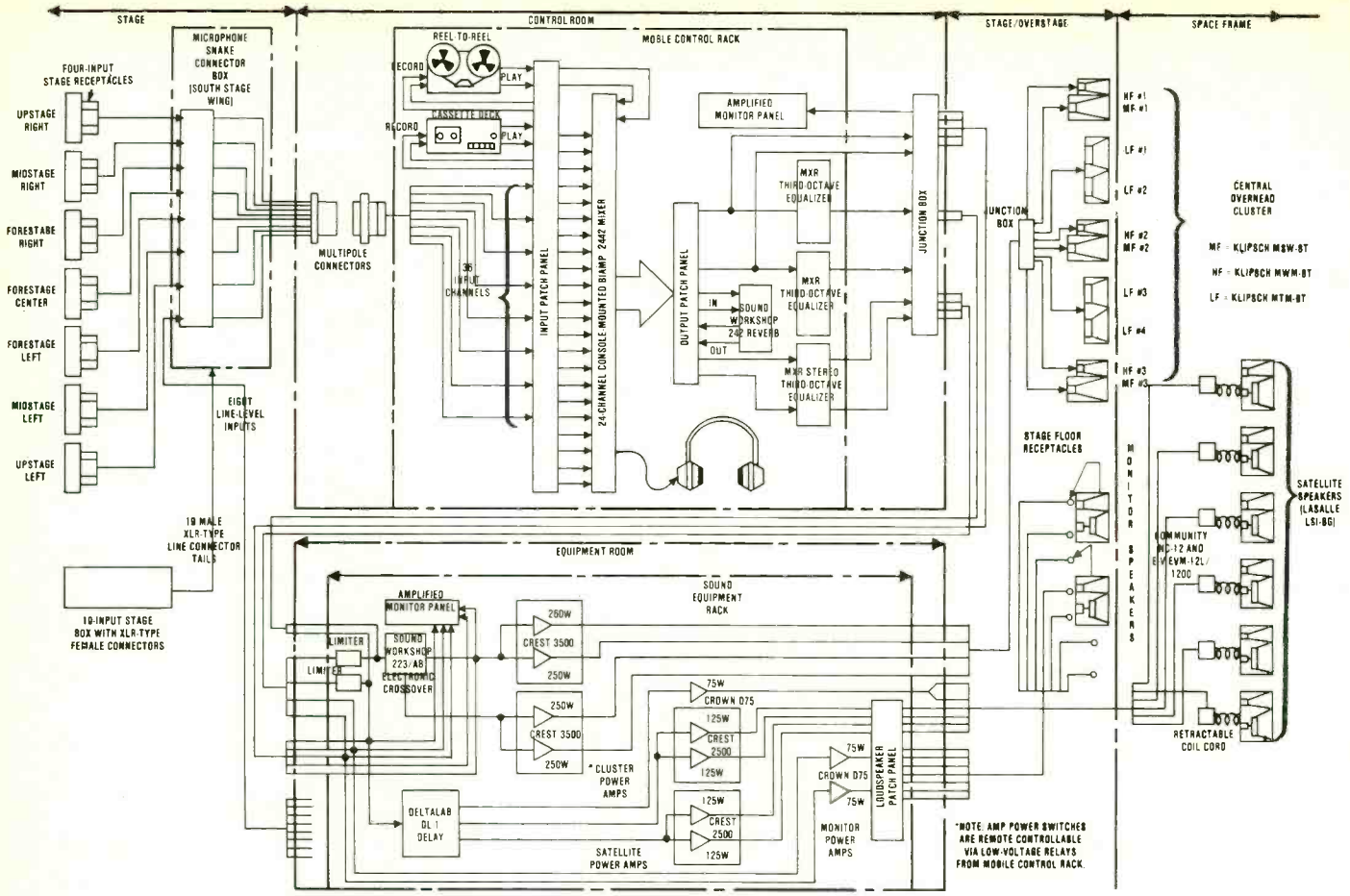

SOUND SYSTEM SCHEMATIC FOR PECK PAVILLION, MILWAUKEE PERFORMING ARTS CENTER

THEATRE SOUND SYSTEM DESIGN

sionally, a bold house management will permit the removal of a half dozen seats (two rows of three) in a plum location just forward of the balcony on the main floor.

System Design

The sound system engineer, having decided on the major components of the system, must prepare drawings and specifications for bid by qualified contractors. Unlike disco or studio systems, there are few people working in the field of theatre sound that can design and build turnkey systems, other than contractors, of course, who also can do their own designing.

One of the most important roles of a sound system engineer is to ensure that the equipment and systems are purchased and installed with all the care with which they were designed. Specifications and drawings have to be very carefully prepared. Very few theatres are privately built these days, and so there is almost always a need for competitive bidding. Oftentimes there are legal reasons why the low bidder must be accepted. Therefore, the bid documents (both specifications and drawings) have to be scrupulously prepared to sort out the bad from the good (experience, reputation, ability), to ensure that the integrity of the design is incorporated within the installation, and to attempt to eliminate quality-eroding short cuts. It is not unreasonable to expect the contractor to be an authorized installer of the amplification that he will install, nor that he has at least five similar projects under his belt, 10 years experience, and the financial ability to complete the job at hand.

After the bid has been awarded, the engineer has to review submittals and shop drawings prepared by the contractor, then inspect the installation during certain critical phases or milestones. When everything is in place, the system has to be tested, equalization checked, and final acceptance tests conducted by the engineer. The testing procedure includes making careful checks of dynamic range, distribution, and frequency response over the full seated audience area. Equalization of the house is done in a specific sequence. Impedances, polarity and phase measurements are made. Critical distances, acoustical gain, percentage articulation loss, and so on need to be determined, and all filter settings recorded. We normally require that the contractor builds into his bid adequate time for instruction of the owner's personnel/operators. This takes place at the time of acceptance, after which the owner signs off the contractor. Twenty-four house service calls and bi-monthly visits are expected follow-up responsibilities under the first year's warranty.

Sound Designer's Preferences

Thus, having completed the design and installation of a state-of-the-art system in the brand new or renovated theatre, the sound system engineer leaves with pride and waits for the first show to load in. On his return, he learns that the liberal policy of the house has allowed the show-sound designer to bring in his own hardware. To his horror, the engineer notes the horizontal column speakers chained to the face of the balcony, hanging down and pointing at the rear wall beneath the balcony. Large monitors are strapped to torm booms (this is theatrical parlance for the vertical pipes held in sandbagged or heavy steel bases like giant microphone stands) on the forestage, pointing every which way. Despite the work put into the acoustical design of the orchestra pit, there are microphones at every music stand. There is more gaffer tape than cable on the floor and walls.

But what about the carefully tailored house system? Aha, therein lies the hint of basic difference between the Sound System Engineer and the Sound Designer: the experienced/calculated versus the experienced/subjective approach. This is not a road show where expediency is paramount, but a show that will run in the house for several weeks. Or at least long enough for the sound designer to be able to adapt his concepts, through experimentation, to house amplification/effects systems that have just been installed at great cost.

Unfortunately, the designer has come into the theatre dragging the legacy of many previous "impossible" houses, and so he is not about to take any chances. The

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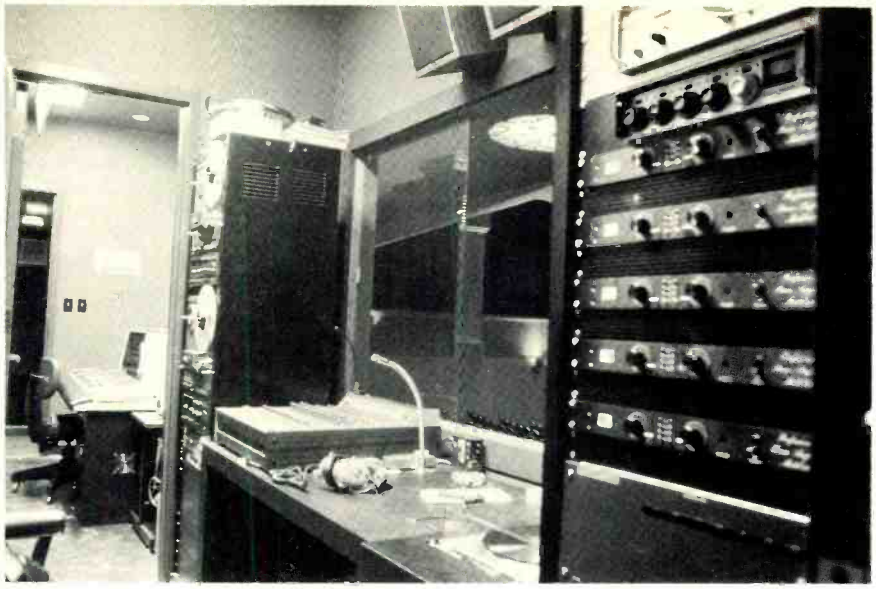


Peter George

Sound console and outboard signal processor racks at the Roger Stevens Center for the Performing Arts, Winston-Salem.

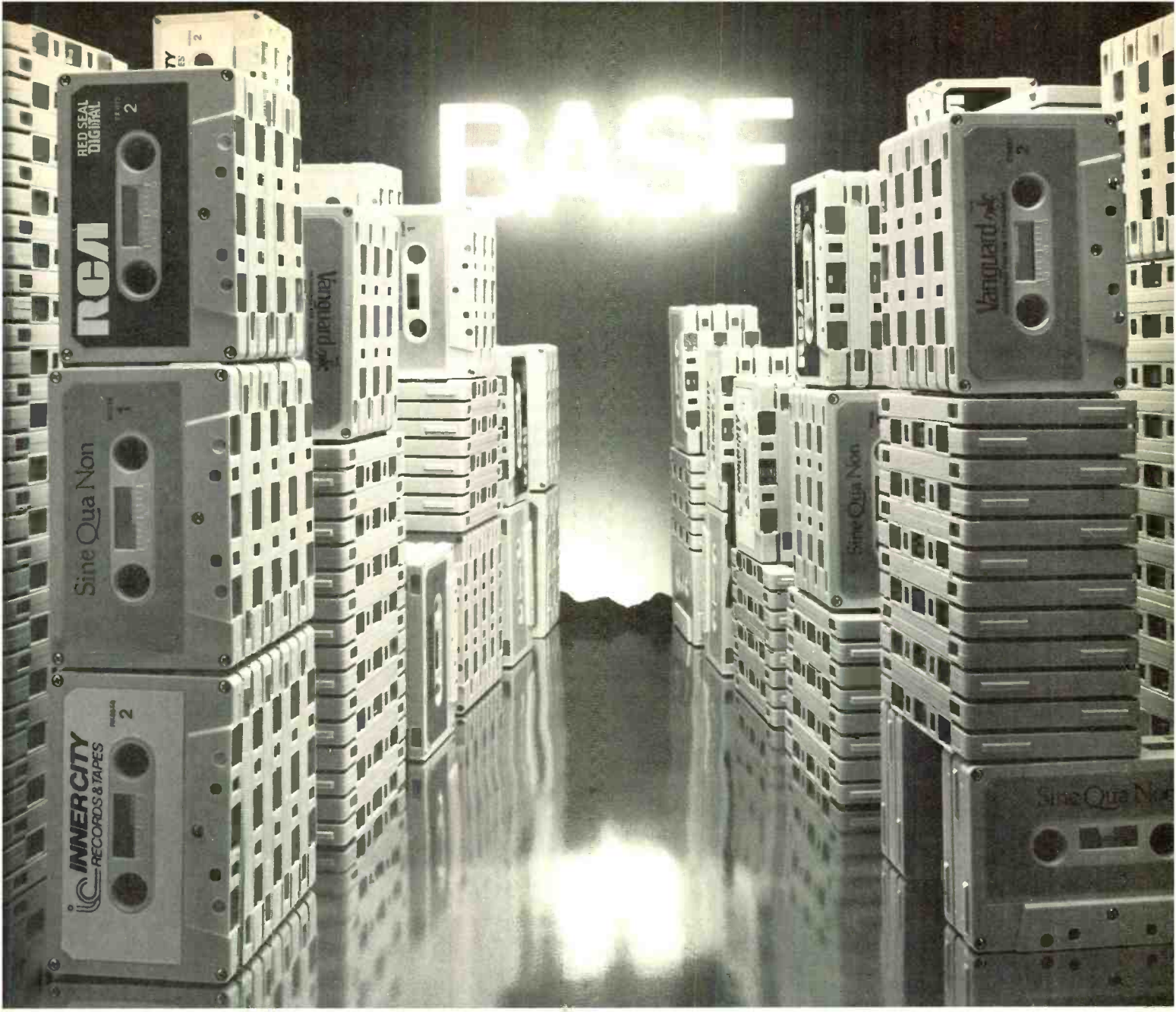
popular conception that sound system engineers and, more pertinent, acousticians, are more often wrong than right is not wasted on him. This even assumes that he is able to find out that the house electronics were professionally designed. In any case, he discovers that the house system only delivers 98 dB SPL to each seat. He needs at least 118 dB to ensure that the show is a success in spite of itself, not to mention the need to compensate for the 20 dB Permanent Threshold Shift he suffers from as a result of 10 years on the road with the Grateful Led Leppards. (Speaking of sound levels, legend has it that in the case of one particular Broadway musical that was ebbing fast, the producer stormed up to the operator dutifully riding gain on the board at the rear of the house, elbowed him aside and cranked those sliders all the way up. The producer evidently believed that the shortcomings of the production could be masked by desensitizing sound level.) Anyway, it is probably only natural that the sound system designer should justify his keep by pouring gentle disdain over the quality of the built-in electro-acoustics.

Too Much of a Good Thing?

Recent screams from audience and critic alike, however, would indicate that things have gone far enough (too far?) in the amplification of shows. This writer also regrets the need to amplify drama that, with the right professionals, good blocking and a quiet audience, need never require electro-acoustics except for Noises Off, Distant Thunder, etc. When plays like *Da* and *Plenty* are amplified, even with Broadway-quality casts, you begin to wonder what is going wrong with live theatre. Now, those of us familiar with Broadway houses know that there are some acoustical rocks among the gems, but these theatres would not have survived all these years if they were not at least acoustically viable (i.e. average).

Unfortunately, I see the proliferation of the sound designer as a development more closely related to fashion than need. Which is fine if it provides some of our colleagues with an income, but when it turns off an audience, I wonder at the wisdom of it all.

A recent, positive innovation in theatre electro-acoustics has been the introduction



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THEATRE SOUND SYSTEM DESIGN

of hard-of-hearing systems. There are a variety of available methods, but the one that seeems to have caught on for reasons of simplicity, cost and ease of installation, is the Sennheiser infra-red system. This employs low-profile emitters at the proscenium wall, and lightweight headsets for the patron that has a hearing impairment. It can also function as a translation system. During a recent visit of a Kabuki theatre company to the Metropolitan Opera, patrons were able to rent headsets for a running commentary on the otherwise somewhat inscrutable action on stage.

Another type of system now available uses FM transmission, and is available from Phonic Ear or Williams Sound. Induction loop systems are occasionally found, although they are more common in Europe.

It will be necessary to include monitoring, paging, closed-circuit TV and production communications systems as part of the overall theatre electro-acoustics package. These systems will not be described here, however, since they are subject mat ter for a complete article in themselves.

David Kahn

Author Peter George carrying out sound performance checks in the control booth of the Auditorium, Scarsdale High School, New York.

Electronic Acoustics: Assisted Resonance and ERES

A review of theatre sound would not be complete without mention of two electronic "assist" systems that have been put into use in some spaces both here in the U.S. and in Europe. The first of these, a system developed by the AIRO (Acoustical Investigation and Research Organization Ltd.) in England, provides Assisted Resonance for those spaces that do not have a long enough reverberation time. Developed in an attempt to enhance the dry acoustics of the Royal Festival Hall, London, the system uses the principle of controlled feedback in a series of closed circuit channels. Each channel consists of a microphone housed in a tuned resonator located in the ceiling, between 72 and 90 of these mikes being employed to send signals back to a central rack where they are phase-adjusted, filtered and then amplified, using microprocessors. The microphone, having received sound energy at a location of peak sound pressure, is coupled to a loudspeaker located at another peak in a loop that is adjusted to be in-phase.

According to AIRO engineers, each channel is made "frequency selective so that any number of channels with a designed frequency separation can be used to modify the reverberation characteristics." Such a system permits a theatre to instantaneously vary its reverberation from, say, 1.3 seconds (suitable for speech), to 1.8 seconds (suitable for opera), to 2.1 seconds (for 19th Century symphonic/romantic music) to 2.7 seconds (for organ or choral music), which has distinct advantages in allowing augmentation of sound in existing dry halls lacking in natural reverberation. It also has the advantage, with the design of a new space, of permitting by architectural means a fixed reverberation, that is on the dry side, while electronically providing a much longer reverberation when desired, than would normally be permissible with a low cubic volume space.

It must be remembered, however, that such a system needs careful integration with (and probable modification of) the surface boundary architecture to incorporate the resonators. It is *not* a system that can just be set up within the house like a sound system. Thus, as a retrofit, it is limited to certain very specific situations.

The same microphones used to supply input to the Assisted Resonance system also can be used to feed signals to a second system known as ERES (Electronic Reflected Energy System), as developed by acoustician Christopher Jaffe. However, the primary signal for this latter system usualy comes from a *single* microphone located above the stage, the outputs being processed through equalizers, delay devices, and amplifiers before being sent to loudspeakers. This augmented sound energy is concentrated in the range 250 Hz

Peter George

Central overhead loudspeaker cluster in ceiling of the Creative Arts Center, Shepherd College, Shepherdstown, West Virginia.

to 6 kHz, and is delivered to each seat within 20 milliseconds (or so) of the direct sound, to simulate early lateral sound where it is not normally available to the audience. In addition, the system can also be designed to deliver low-frequency energy with a delay of as much as 100 to 200 milliseconds to provide greater warmth. [The AR/ERES systems installed at the Eugene Performing Arts Center, Oregon, were described in an article published in the December 1982 issue of *R-e/p — Editor.*]

In theory, AR and ERES systems provide the acoustician with additional tools that are of particular value under certain circumstances. However, there is still massive resistance by the musical community (in particular) to anything that hints of electronics, so that the purveyors of such systems are generally treated as no better than snake-oil salesmen. There is, nevertheless, good reason for concern where the human element is concerned. If you have ever had to re-equalize a house after a hobbyist had pried off the plate to "juice up" certain frequencies that he found lacking, you will quake at the thought of what damage could be done at the console of an AR/ERES combination. Fortunately, unless the computer program is damaged, this can quickly be put right. But the point is that with such control at his fingertips, a deranged operator can reset the house acoustics at whim to its terminal detriment. In spite of this, I believe that both systems are very useful tools to have available. We welcome their appearance and their growing acceptance on the scene.

As with all electronics there are budget systems and there are gold-plated Cadillacs. However, as a guide, the type of budget that we should consider reasonable for a small theatre would be between $85,000 and $100,000, installed. Of course, the gold-plated budget is limitless, but a well-equipped performing arts theatre should allow between $300,000 and $400,000. This would not include any exotics such as Assisted Resonance or ERES, which need to be budgeted on a job-by-job basis. ■■■

Author's Note: I have mentioned a variety of products and manufacturers in the above article to better identify the various types of components in the systems described. This should in no way be taken as an endorsement of these products, nor should the omission of other names be taken as a lack of endorsement — *PG.*

STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

Northeast:

☐ **MAYFAIR RECORDING STUDIO** (New York City) has been acquired by **Jay Siegel**. The former producer for Don Kirshner Entertainment Company says that he plans to make major capital improvements to make the facility "nothing short of a first-class studio operation in every way." In addition to extensive decorative efforts throughout the complex, Siegel will undertake a comprehensive upgrading of outboard equipment, the installation of an automated MCI JH-636 36/24 console in Studio A, and the addition of video interlock and fullcoat mag transfer capabilities. *701 Seventh Ave, New York, NY 10036. (212) 581-2128.*

☐ **CLASSIC SOUND** (New York City) is a recently established 24-track studio located in Manhattan's Soho district, and equipped with an MCI JH-114 24-track with Dolby, Tangent 3216 30/24/16 console, Ampex AG440-C two-track, plate reverb, and a large selection of mikes including a Neumann U-47 tube, and RCA ribbons. Studio facilities include a Steinway B grand piano, drum and vocal isobooths, plus a three-way cue system. *548 Broadway, New York, NY 10012 (212) 925-1839.*

ICE ASSOCIATES — Major Upgrade

☐ **ICE ASSOCIATES** (Bala Cynwyd, Pennsylvania), a radio and TV jingles production facility, has upgraded with the addition of a Yamaha Model 1608 console, Otari MX-5050 Mk III 8- and two-track machines, a Yamaha D-1000 digital reverb, DeltaLab Effectron II digital effects, and a Yamaha DX-7 synthesizer. *225 City Line Avenue, Bala Cynwyd, PA 19004 (215) 668-0714.*

☐ **UNIQUE RECORDING** (New York City) recently added two Otari MTR-12 half-inch two-tracks for its editing room; 10 Onkyo 2070 cassette decks for dupes; a second Ursa Major Space Station, six Valley People Kepex II gates and four Gain Brain IIs for Studio B; and eight Neve Modules for Studio A. In the synthesizer corner a PPG 2.3 and Waveterm, a Juno 106, Korg Poly 800, and the Roland GR700/707 Guitar synthesizer controller are now available for use on sessions. *701 Seventh Ave., New York, NY 10036 (212) 921-1711.*

☐ **EASTERN ARTISTS RECORDING STUDIO** (East Orange, New Jersey) has added an AMS RMX digital reverb unit, and four new Valley People Kepex II noise gates, bringing the available total to eight. **Pam Haynes** is the newest staff addition, taking the position of assistant manager. *36 Meadow Street, East Orange, NJ 07017. (201) 673-5680.*

☐ **STUDIO 4** (Philadelphia) has completed an upgrading of its 24-track studio and, through the services of **Video Rock**, will offer full video facilities. New equipment includes a Studer A80 MkIII 24-track and Studer half- and quarter-inch machines. New outboards include a Lexicon digital reverb and Prime Time, another Eventide Harmonizer, and three more UREI LA-3A limiters. The new console is a custom Neotek Series IIIC 38/24 with sub grouping. Video Rock's facility includes a video studio, video switching with chroma key and effects, and ¾-inch shooting and computer editing. Video Rock will be primarily involved in the production of music videos. *444 N. Third Street, Philadelphia, PA 19123. (215) 923-4443.*

☐ **KAMPO CULTURAL CENTER** (New York City) has begun construction of a 1,200-square-foot studio facility in downtown Manhattan. Scheduled to open in September, the studio has been designed by **Maurice Wasserman Associates** with help from **Al Fierstein** of Acoustilog. Recording hardware includes a 28/24 Sound Workshop Series 30 console; Otari MTR-90 24-track; MTR-12, Ampex ATR-102 and Revox A-700 mastering decks; UREI 813A, 811B, JBL 4312, and Auratone 5C monitors; plus outboards from MicMix, Furman, Lexicon, Aphex, Orban, UREI, Audio+Design/Calrec, Eventide, and dbx. A video production studio, modular theater, and gallery are currently being designed for the premises, and will be available for use early next year. *31 Bond Street, New York, NY 10012. (212) 228-3063.*

☐ **NIBOR RECORDING STUDIO** (Hurley, New York) formerly Ontrax Recording, has upgraded from 8- to 24-track with the purchase of a Soundcraft 1600 Console and a Soundcraft SCM-760 24-track. In addition to redesigning the entire studio, the facility has also acquired a Studio Technologies Ecoplate reverb system, Lexicon Prime Time and Prime Time 2, and Symetrix compressors and noise gates. Mastering decks include an Otari MX-5050B2 half-track, and a Sony PCM-F1 digital audio processor. Coming soon to the studio: a new Kurzweil 250 digital keyboard. *Hurley Mountain Road, Hurley, NY 12443. (914) 331-3060.*

☐ **EUPHORIA SOUND STUDIO** (Revere, Massachusetts) has upgraded to 16-track operation, according to owner **Howard Cook**. The new installation, which was designed and carried out by chief engineer **Gordon Hookailo** and staff, includes a newly designed control room, equipment racks, and stereo cue system. New equipment includes an Otari MTR-90 16-track, with remote control and autolocator, as well as a new Soundcraft console. *90 Shirley Ave., Revere, MA 02151. (617) 284-9707.*

☐ **PRESENCE STUDIOS** (East Haven, Connecticut) has installed a new Studer A-80 Mk III multitrack with 16- and 24-track headstacks. Other additions include: two additional Hafler DH-500 power amps to tri-amp the studio's Westlake Audio HR-7 monitor system; Yamaha NS-10 monitors; another dbx Model 903 compressor/limiter; a Lexicon Super Prime Time; another Neumann U-87; and five more pairs of AKG K-141 headphones. *461 Main Street, East Haven, CT 06512. (203) 467-9038.*

☐ **KAJEM RECORDING** (Gladwyne, Pennsylvania) continuing its $100,00 update, has added an EMT 251 digital reverberator, Pultec equalizers, Teletronic LA-2A limiters, AKG and Neumann tube mikes, and and Oberheim DMX drum synthesizer. *1400 Mill Creek Road, Gladwyne, PA 19035. (215) MIX-EARS.*

☐ **CELESTIAL SOUNDS** (New York City) has installed a collection of vintage AKG and Neumann tube microphones, Neve equalizers and pre-amps, as well as an EMT 140 stereo reverb plate, according to studio manager **Dennis O'Donnel**. *919 Second Avenue, New York, NY 10017. (212) 355-4825.*

☐ **BART CURTIS RECORDING SERVICE** (Elmira, New York) is the new name of Revelation Recordings. According to **Bart Curtis**, president/owner, the new name also signifies a new production format — a mobile unit instead of a fixed location. Utilizing a modified 23-foot Welles show trailer, Curtis says he moved his entire operation, including a 16-track Tascam 85-16 with dbx noise reduction, a RAMSA Model 8816 mixer, and UREI parametric EQ. *424 West Gray Street, Elmire, NY 14901. (607) 732-0950.*

South Central:

☐ **SOUNDS UNREEL** (Memphis), a new two-room studio complex owned by **Don Smith** and **Jon Hornyak**, opened its doors April 1st. According to **Murphy Odom**, owner of Phase Audio, a Memphis-based sound-contracting firm, the design incorporates economy and flexibility for various recording projects. Studio A offers 24-track capability and will be used primarily for record projects, while Studio B has eight tracks, with the capability for expansion, and will be used for voice-over work. Assisting in the design of the two studios was Nashville acoustician **Steven Durr**. Phase Audio provided all the equipment for the studios including customized monitors which were designed by Durr. *Memphis, TN.*

STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

☐ **OMEGA AUDIO** (Dallas) has added **Curtis Drake** as operations manager. Prior to joining Omega, Drake was a staff engineer at Cherokee Studios, Hollywood, and previously worked as a freelance engineer for numerous studios in West Germany. "This business has become very technical in recent years," stated **Paul Christensen**, Omega's President, "and because of this trend, people are our most important product. Curtis Drake is a 'people' professional, well steeped in technology, who will be an invaluable asset to our company." *8036 Aviation Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

☐ **MUSCLE SHOALS SOUND STUDIOS** (Sheffield, Alabama) has installed a Mitsubishi X-80A digital two-track in Studio B to complement its NECAM-equipped Neve 8088 mixing console, according to studio owner **Jimmy Johnson**. "I think everybody is watching the new technology," says Johnson, "and we don't want to be left behind." Johnson cites two major reasons for choosing the X-80A: its accurate sound reproduction, and razor-blade editing capability. "The digitally recorded sound is almost a perfect mirror image of the multitrack; it's like playing the multrack back when I hear the digital two-track version. And the fact that the tape can be cut and spliced with a razor blade like conventional analog tape makes the editing process so much easier for everybody." Muscle Shoals' first project to be completed using the X-80A is a single for George Michaels of Wham U.K. "Right now, most acts are using a half-inch machine as back-up for the Mitsubishi X-80A digital," Johnson points out. "But every artist I've seen here has always gone with the digital master." *P.O. Box 915, Sheffield, AL 35660. (205) 381-2060.*

☐ **GROUNDSTAR LABORATORY** (Nashville) has purchased two new Studer A800 MkIII 24-tracks to replace Studio A80VU recorders, which were taken in trade by Studer. The A800s will be used with SMPTE interlock for 46-track recording capability. *12 Music Circle South, Nashville, TN 37203. (615) 244-4861.*

☐ **THE BENNETT HOUSE** (Franklin, Tennessee) has installed a pair of new Studer A800 MkIII 24-tracks. The two-room facility, owned by **Norbert Putnam** and **Bob Montgomery**, has also recently acquired a Lexicon 224 digital reverberation system with LARC. *134 4th Avenue North, Franklin, TN 37065. (615) 790-8696.*

Mid West:

☐ **MULTI-TRAC RECORDING AND PRODUCTION STUDIOS** (Redford, Michigan) has upgraded with the addition of an API M3000 transformerless console which has 24 input and 24 output channels, six stereo effects sends, and 10 echo returns. Also added is a 3M M79 24-track with autolocator and automatic input switching. According to studio president **Nick Canzano**, the studio has also added a new client lounge complete with two UREI Time-Align monitors. *25533 Five Mile Road, Redford, MI 48239. (313) 531-5333.*

☐ **THE GHL AUDIO ENGINEERING TRUCK** (Hillard, Ohio), which features acoustics and systems design by **Gary Hedden**, is housed in a 40-foot Great Dane trailer equipped with a tandem, air-ride suspension, and enclosing more than 2,800 cubic feet of space. Interior space is divided into three main sections: power and HVAC; control room; and maintenance. A combination wired/wireless intercom system by Clear-Com provides two-way communication and paging for up to six truck crew members, plus direct access to existing house or video intercoms. A high-resolution CCTV is fully remote controlled from the truck. A 300-foot multipair audio cable provides a minimum of 36 balanced microphone or line inputs, and up to 12 balanced line outputs. Microphone inputs may be 48-volt phantom powered, and/or fully transformer floated from house sound equipment via Jensen transformers. The primary audio console, a custom-configured Harrison MR-4 with 36 input/output modules, ARMS automation, 36-segment PPM/VU metering, and phase meter, is equipped with 24 output busses, four auxiliary sends and returns, a stereo foldback system, two independent stereo mix busses, two control room monitor feeds, and studio playback/talkback. The automation system provides input grouping and tape-based fade automation. Tape machines comprise two Otari MTR-90/II 24-tracks, an Otari MTR-12-I four/two-track, two Otari MX-5050-B stereo decks, two Aiwa AD-F990 cassette recorders, and a JVC CR-6650 U-Matic VCR (the latter to use with a dbx Model 700 CDPM processor). A BTX Shadow system with Softouch controller is set up to handle all transport and editing functions. Outboards include two Lexicon Model 200 digital reverbs, two Lexicon PCM-42 digital delays; an Eventide H910 Harmonizer; four dbx 903 compressors, two 902 de-essers, and a 906 flanger/delay; six Valley People dynamics processors; four Aphex EQF-2 equalizers; and six Aphex CX-1 compressor/expanders. Monitor speakers and headphones are driven by Hafler Pro-Series power amps. Control room speakers are Fostex LS-3, Fostex G-7000, and Auratones. Studio playback and cue is via Modular Sound TA-12 speakers and Fostex T-20 headphones. Rates are $2,750 per day ($10,000 per six-day week), which includes one engineer and two assistant engineers, plus $1 per mile portal-to-portal from Columbus, Ohio. *P.O. Box 273, 3999 Parkway Lane, #13, Hillard, OH 43026. (614) 876-1057.*

GHL AUDIO — New Multitrack Mobile

☐ **ALPHA RECORDING** (Lombard, Illinois) has installed a new Sound Workshop 32/24 Series 34 console with ARMS automation. New monitors are Tannoy SRM15X powered by a Hitachi Class G 200W amps. In addition to the Tannoys, the facility has added a pair of Yamaha NS-10M speakers for smaller speaker comparison. The staff has extensively remodeled both the studio and control room. **Bobbie Thomas** is Alpha's owner, and **Corinne Karpiak** its manager. *515 West Harrison, Lombard, IL 60148. (312) 495-2241.*

TRADE SECRET — Custom Complex

☐ **TRADE SECRET RECORDING** (Cleveland) recently opened a new 2,200-square-foot, custom designed recording complex. Highlights include the hugh variable acoustic studio space with high ceiling, hardwood flooring, full floating vari-trap iso booths, live cell, and isolation room. Full-scale monitoring is built into the the live end. the control room comprises an advanced "EDEL acoustic transformer design" with JBL 4350 monitoring. Featured equipment includes an MCI 16 track, Neotek Series III console, and an EMT 251 digital reverberation system. *9407 Olde Eight Road, Northfield, OH 44067. (216) 463-0646.*

☐ **STUDIO A** (Dearborn Heights, Michigan) has added a Sony PCM-F1 digital audio processor and Lexicon Model 200 digital reverb. With the addition of the F1 to the New England Digital Synclavier II synthesizer, Studio A describes itself as moving even further in the direction of digital recording. *5629 N. Beech Daly, Dearborn Heighs, MI 48127 (313) 561-7489.*

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STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE

☐ **AUDIO GRAPH PRODUCTIONS** (East Lansing, Michigan) has updated its 24-track facility with the addition of a Lexicon 224X digital reverb with LARC, Lexicon Prime Time II, Aphex Exciter, Eventide Instant Flanger, additional Valley People Kepex, UREI compressors, Neumann and AKG microphones. *2810 Bennett Road, East Lansing, MI 48864. (517) 332-3272.*

☐ **ARS RECORDING STUDIO** (Chicago Heights, Illinois) owned and operated by **Gary Cobb** of ARS Enterprise, has purchased the Timbre Recording Facility in Alsip, Illinois. ARS will relocate its Chicago Heights studio to this new location. Recent purchases for the new ARS location include a Trident 28/24 Series 70, 3M 16-track, UREI 811B monitors, plus Lexicon Super Prime Time and Ursa Major 8 × 32 digital reverberation units. *268 West Normandy, Chicago Heights, IL 60411. (312) 754-9191.*

☐ **PINEBROOK RECORDING STUDIOS** (Alexandria, Indiana) recently completed a comprehensive expansion and renovation program. The facility now has three studios in operation, offering a full range of recording options from voice-overs to audio/video orchestra sessions. Designed by audio architect **John Edwards**, and acoustically tuned by **George Augspurger**, the new Studio C is equipped with a 44-input Quad Eight Coronado Series, fully automated console featuring discrete circuitry. The 2,600-square-foot studio area features two isolation booths and a raised platform. Other features include: 48-track recording and mixing; Ampex ATR-100 two- and four-track machines; MCI JH-24 24-track; UREI 813-A Time-Align monitors; Hafler and Crown power amps; Studio Technologies Ecoplate and digital echo, as well as a wide array of outboard gear, mikes, special effects and auxiliary monitors. The 1,500 square-foot Studio A is equipped with a fully automated 32-input MCI JH-636 console, MCI JH-24 24-track, two Ampex ATR-100 two-tracks, EMT-140 and EMT-245 digital echo, UREI 813-A Time-Align monitors, and features a raised rhythm alcove and isolated drum booth. The third studio in the complex, Studio B, is used primarily for editing, copying, program put-togethers, voice-overs, and client/studio demos. To meet the specific needs of its clients, the studio offers a host of synthesizers, including a Yamaha DX-7, Prophet 5, ARP Odyssey and ARP string ensemble, Yamaha drums and LinnDrum. *P.O. Box 146, State Road 9 South, Alexandria, IN 46001. (317) 724-7721.*

Southern California:

☐ **OAK RECORDING** (Anaheim) has established a joint operating venture with **calMedia**, a full-service advertising and production agency. The new venture will be know as **calMedia Recording Services**. Oak will function as an in-house operation continuing to offer a multitrack recording studio, on-staff composers, arrangers, and producers, conducting record manufacturing and other recording services. These changes will enable the new company to offer film scoring, jingles, media placement, slide and tape presentation and sync time-coding. Plans are underway to construct a 1,500-square-foot film-scoring studio, with 16mm, 35mm, and VCR projection capabilities. *440 South Anaheim Blvd., Anaheim, CA 92805. (714) 520-3121.*

☐ **SOUND SOLUTION** (Santa Monica) recently completed a six-month studio expansion that includes a complete upgrade of the facility, as well as the acquisition of new studio equipment, including a Lexicon 224-XL digital and EMT-140 plate reverb system, an MCI console and 24-track, a Yamaha grand piano, the new Yamaha DX-7 digital programmable synthesizer, a LinnDrum

computer, and a large selection of new and vintage microphones. Sound Solution specializes in master and demo recording, television, film, radio and theatrical production. *1211 Fourth St., Santa Monica, CA 90401. (213) 393-5332.*

☐ **CALIFORNIA RECORDING** (Hollywood) recently upgraded its reverberation system with a Lexicon 224x with LARC, and two Lexicon PCM-42 digital delay lines with four-second memory option. Also purchased were four Valley People Kepex II noise gates. *5203 Sunset Blvd., Hollywood, CA 90027. (213) 616-1244.*

KDISC's senior cutter KEN PERRY

☐ **KDISC MASTERING** (Hollywood) has appointed **Ken Perry**, formerly with Capitol Records, and most recently a partner in Masterfonics, Nashville, to the post of senior cutting engineer, according to VP **Bill Lightner**. "Kdisc offered me the perfect opportunity to return to Hollywood and I've been re-establishing contacts with my former clients here," Perry explained. A 15-year veteran in the mastering industry, he left Capitol in 1982 to become a partner and VP of Masterfonics. "I needed a change after 13 years and I wanted to have more control over my working environment. But I really missed California, and Kdisc gave me the perfect reason to come home." According to Lightner, "The appointment of Perry to our staff is one reflection of the many changes we are making in 1984 to increase recognition of Kdisc in the industry. We are also adding a third complete cutting room and renovating our building." *6550 Sunset Blvd., Hollywood, CA 90028. (213) 466-1323.*

Northern California:

☐ **PATCHBAY STUDIOS** (San Rafael), to compensate for room and speaker coloration, has installed new Meyer Sound 833 Studio Reference Monitors. "The power, clarity, and especially the unbelievable accuracy of the 833's are unparalleled by anything in our experience," say studio staff. *2111 Francisco Blvd. #7, San Rafael, CA 94901.*

☐ **TRES VIRGOS STUDIOS** (San Rafael) has named **John Clavin**, formerly chief technician with AIC Sales, Los Angeles, and Fantasy Studios as chief technician and staff engineer. Clavin's areas of responsibility will extend into the area of research and design in "the studios' never-ending search for the last decibel." **Steve V. Johnson** now serves as the studios' operations manager, handling client relations, inventory and sales, as well as assistant engineering. Also, **Mary Ann Zahorsky**, computer and music student at UC Berkeley and Stanford, has joined Tres Virgos as a staff assistant, while **Stuart Hirotsu** recently was promoted from staff assistant to assistant engineer. *1925 Francisco Blvd., San Rafael, CA 94901. (415) 456-7666.*

Great Britain:

☐ **TELEVISION SOUTH WEST** (Plymouth) has become the first major UK broadcasting company to order a new Studer 900

mixing console. The model ordered is a specially developed "Hexmix" version of the 902/8/2, in which Studer has managed to provide 24-track monitor mixing within a very limited space. **F.W.O. Bach**, UK agents for Studer and Revox equipment, is supplying the console as part of an overall contract for the new TSW dubbing suite, which also includes a Studer A80/VU multitrack, a Studer A810 two-track with center timecode, ITC cartridge machines, and EMT turntables. *Plymouth, Devon.*

☐ **EMI/ABBEY ROAD** (London) recently installed a Neumann VMS82 Direct Metal Mastering disk cutting system. The Teldec DMM technology involves cutting into copper for direct production of the stamper, thus eliminating the numerous processing steps involved in conventional lacquer mastering. The new installation is said to be the first DMM suite in the UK, and reflects not only EMI Record's worldwide commitment towards DMM, but also Abbey Road's consistent policy of technical advancement. Already, experimental cuts are said to be demonstrating the improvement of cut, and faster factory turn-around. Another major record company, Virgin Records, has also shown its commitment to DMM by ordering a VMS82 DMM for a cutting suite at Townhouse Studios, West London. *3 Abbey Road, London NW8 9AY. (01) 286-1161.*

Ireland:

☐ **WINDMILL LANE STUDIOS** (Dublin) has taken delivery of a new Solid State Logic SL4000 E-Series Master Studio System, with studio computer and plasma bargraph metering. The facility, designed by **John Storyck**, is described as the first Irish studio to equip itself to meet international expectations. It was created by **James Morris**, who formerly had a film editing business, and **Brian Masterson**, who had been a musician and freelance recording engineer. The team is completed by **Russ Russel** and **Meiert Avis**, both film editors. As well as two sound studios, Windmill Lane has facilities for broadcast video post-production, film editing, and off-line editing. *Dublin, Ireland.*

AROUND THE STUDIOS:
A Digital Spotlight of Southern California Facilities

•Terry Williams, studio manager at Kenny Rogers' **Lion Share Studios**, Hollywood, reports an increase in business since installing Mitsubishi X-80 two-track and X-800 32-track machines. "For me, just generating a cash flow isn't enough," he says. "Generating a list of respected clients is what's important."

Some of the artists and producers that have been using the digital mastering system include: producer Jimmy Iovine doing the movie soundtrack album for *Streets of Fire*; Phil Ramone preparing Barbra Streisand's *Yentl* tracks for album release; Michael Sembello's *Automatic Man*; Jeffrey Osborne's "Stay With Me Tonight," produced by George Duke, as well as all of Duke's projects with Stanley Clarke; Donna Summer's "She Works Hard for the Money," and Rod Stewart's newest album, with Michael Omartian producing both artists; and the Julio Iglesias/Willie Nelson duet "To All the Girls I've Loved."

•**The Burbank Studios**, home of Warner Brothers and Columbia Pictures, has purchased a Mitsubishi X-800 digital 32-track for Scoring Stage 1. According to post-production sound director Tom McCormack, the X-800 will be used primarily to record musical soundtracks for the studio's motion-picture product.

"We want our recording to sound as close to reality as possible," he says. "The Mitsubishi X-800 provides better frequency response by operating at a higher digital sampling rate of 48 kHz. When we play back the Mitsubishi digital, we get almost an exact replica of the original performance."

In addition to 40 interleaved data tracks — 32 digital audio plus eight parity — the X-800 has five dedicated tracks to accommodate SMPTE timecode, analog audio, and other control information. Audio, video and film machines can be locked to the multitrack without sacrificing any of the 32 audio tracks.

"Basically we feel that digital is the wave of the future," McCormack concludes, "and the Mitsubishi X-800 will give us the chance to explore a lot of innovative possibilites down the road."

•At **Oasis Recording Studios**, Universal City, owner Giorgio Moroder has been putting the facility's three new Sony PCM-3324 digital 24-tracks through their paces during scoring sessions for a new soundtrack to Fritz Lang's 1926 silent classic, *Metropolis*, to be released this summer. Moroder purchased the rights to the film two years ago, and has added rare footage previously thought to have been lost.

"I don't know how I could have completed the project without digital," he comments. "There have been so many changes in the past two years that called for multiple generations and re-recording which, in analog, would have created a real nightmare."

The Oscar-winning composer enlisted the talents of such artists as Pat Benatar, Adam Ant, Freddy Mercury of Queen, and Bonnie Tyler, for the new soundtrack. (A glimpse of the film can be seen in Queen's recent music video, "Radio Gaga," where the band is superimposed over the original footage.)

Oasis has three control rooms, each equipped with a PCM-3324 digital multitrack. For the *Metropolis* project, two multitracks were synchronized for the film-scoring sessions.

"We have a library of un-mixed 24-track digital masters," explains Oasis chief engineer Brian Reeves. "These are mixed down to sets of four tracks on the second PCM-3324. Both machines are run in sync with a video workprint. The music mixes end up on adjacent four-track groups that can be shifted [via SMPTE timecode offsets] to correspond with editing changes in the film. The different source material is then crossfaded and recorded directly to the four-track film master for Dolby Stereo in theaters."

The use of digital technology has permitted techniques that would be prohibited in analog, Moroder says. "For instance, partial re-mixing is now possible. Let's say I do a mix and discover that the melody is not loud enough in the second verse. With digital you can drop in at one point in the middle of a mix, make your adjustments, and then drop out. The electronic edits are instantaneous and completely inaudible. Dropping in generally works with analog, but dropping out has always been the real problem."

Reeves, who has worked with Moroder on film-scoring projects for *Scarface* and *D.C. Cab*, comments that there are perceptual changes caused by recording digitally: "It was a strange realization, but digital seemed to sound better as the days went by. With analog, most of the degeneration of audio occurs in the first 48 hours. You

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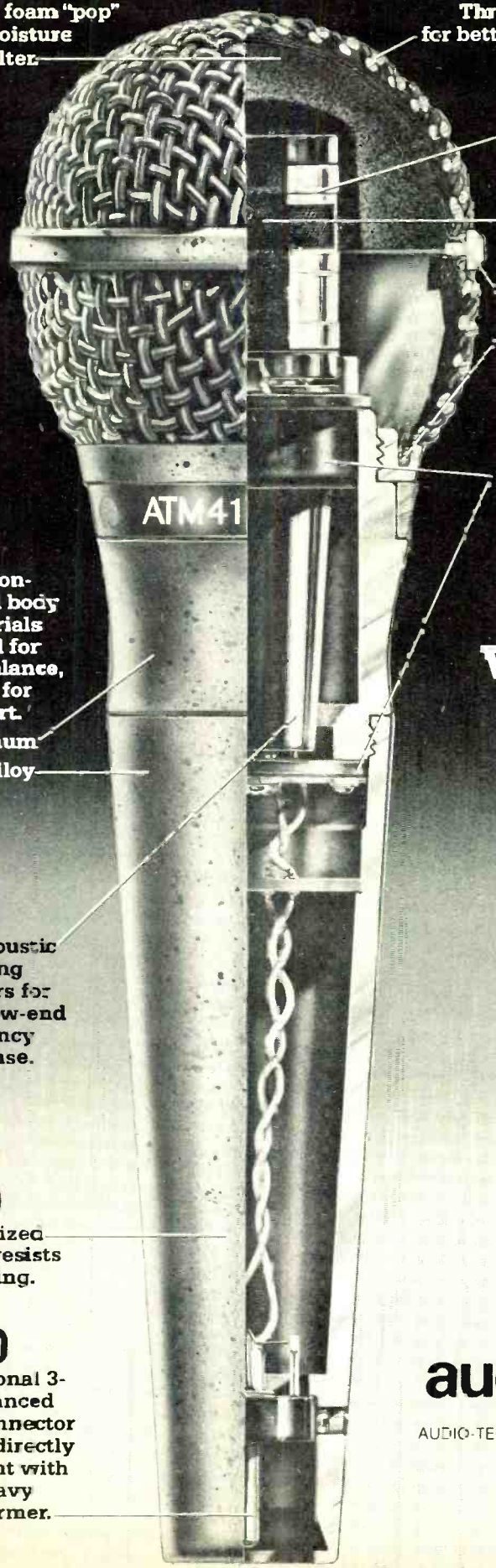

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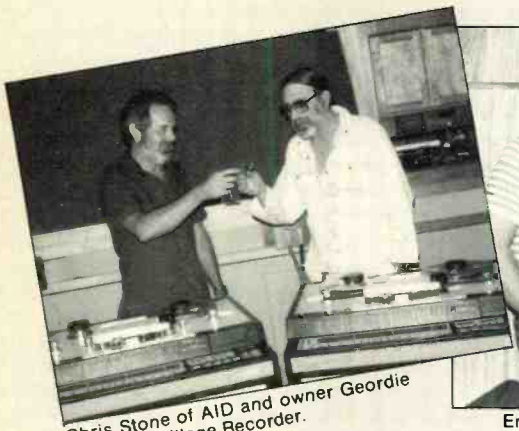
STUDIO FACILITIES EQUIPMENT PEOPLE UPDATE



Chris Stone of AID and owner Geordie Hormel at Village Recorder.

Engineer John Guess and producer Michael Omartian at Lion Share.

Engineer Brian Reeves and Giorgio Moroder at Oasis Studios.

get used to that happening and accept it as normal. With digital, the sound stays exactly the same as when it was recorded — that's why it appears to get better."

In addition to his digital film-scoring work, Moroder feels strongly about the future of the Compact Disc. "I personally think that the CD will take over — it may take a few years, but the sound quality, the size, and the programming features give striking advantages over analog records and cassettes. If someone buys a CD player, then the recordings should be *completely* digital. Somebody has to start doing purely digital recordings, and I'm pleased to be one of the first."

•**Village Recorder**, West Los Angeles, has taken delivery of two Sony PCM-3324 digital multitracks for use in Studios B and D, although the new decks also will find applications in Studio A for tracking dates. According to executive director Michael Geller, purchase of the two new digital multitracks, in addition to the studio's present Mitsubishi X-80A stereo machine, and a Mitsubishi X-800 32-track scheduled for delivery at the end of May, reflects Village's decision to move from being just a "recording studio," to a full-service post-production facility. "We decided to make the commitment to digital," he says, "because it represents the highest state of the art in recording technology for records, film scoring, and audio-for-video. We intend to diversify into related areas of production, including soundtrack scoring — witnessed by our recent setting up of Village/MRI in Hollywood — and video and film post production. Digital allows us to maintain the highest level of audio quality no matter what the project."

The total number of Sony PCM-3324 equipped studios around the country currently runs to 17: two at Neil Young's Broken Arrow Ranch; three at Giorgio Moroder's Oasis Recording Studios; two at Village Recorder; two with John Moran's Digital Recording Services; two at Atlantic Studios, New York; one each at Stevie Wonder's Wonderland Studio, Hollywood, Frank Zappa's facility, Record Plant, Los Angeles, and Power Station, New York; plus two on order for Soundworks Digital Audio/Video Studios, New York.

•**Future Disc Systems**, one of Hollywood's newer disk mastering facilities, reports increased business since the installation of a Mitsubishi X-80A digital two-track, and companion DDL-1 Digital Preview Unit.

"We try to run a facility that gives a client what they're looking for, not what *we* think they should have," says Steve Hall, Future Disc's veteran cutting engineer.

The reel-to-reel X-80A with fixed-head design features traditional analog capabilities like cueing and razor-blade editing. "Studio-wise," Hall offers, " I think the Mitsubishi system is the best way to go, because it lets the engineer work as he has for years without expecting him to change his approach."

The DDL-1 Preview Unit supplies four channels of audio (stereo "program" and "preview" signals) to Future Disc's Cybersonics/Ortofon cutting system, and offers variable delay from 10 milliseconds to 1.9 seconds.

Of course, the primary advantage in mastering from digital, Hall says, is the sonic quality and the excellent noise specs, which complement the facility's other equipment, including its custom-built monitors and automated Cybersonics DRC-202 console which memorizes and controls level changes and outboard gear during a complex cut.

"Mastering is really the last chance to make any changes in the final product," Hall continues, "and we want to interpret and reproduce the client's concepts as accurately as possible."

Future Disc also was used recently to master an all-digital recording project. As reported in the February issue of *R-e/p*, late last year Jerry Barnes, engineer and co-producer, conductor Dr. Gary Bonner, dean of the school of music at Azusa Pacific University in California, and co-producer Don Neufeld went to England to record the Concertante Ensemble of London on a Mitsubishi X-800 digital 32-track at EMI/Abbey Road's Studio 1. The trio then return to the US to complete the recording at United Western Studios in Hollywood, where the Azusa Pacific University 120-voice chorale was overdubbed on a second Mitsubishi X-800, and the final editing and mixdown made to a Mitsubishi X-80 digital two-track.

Hall reports that the mastering session was "pretty straight ahead," with absolutely no set-up or operational problems. "But I had to be very critical with my setting and adjustments," he says. "The subtleties show up so much more easily during digital mastering than on an analog project."

•**United Western Studios'** new Mitsubishi X-800 digital 32-track is said to be portable enough to be moved between any of the company's three studios, according to general manager Jerry Barnes.

"Many of the engineers and producers who work here at United Western need the extra tracks that the Mitsubishi provides," says Barnes. "With all the sophisticated techniques being used today to make records and soundtracks, 24 tracks is no longer enough."

"The punch-in/punch-out capability of the Mitsubishi X-800 is shockingly fast, too," Barnes points out. "We've been able to punch-in and -out of situations that we would never have attempted in analog." ■■■

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Photography by Steven Barnett and Jeff Di Perna

CONAN THE BARBARIAN

Computer Controlled Audio Production At the Universal Studios Tour

by Adrian Zarin

The lights dim as dramatic music swells. A deep, commanding voice invites you to journey back to a time long ago: "The Age of Swords and Sorcery . . . !"

While it may evoke the shadowy recesses of a mythical past, Universal Studios' live extravaganza, *Conan the Barbarian*, represents a futuristic union of computer technology, digital audio and an arsenal of special effects that includes lasers, pyrotechnics, dry-ice smoke, trick elevators and lots of swashbuckling sword play. Now entering its second year, the show has become a crowd-drawing mainstay of the famed Universal Studios Tour.

The extravaganza was conceived and is currently being produced by Peter Alexander, an entertainment industry veteran whose background includes a term as a graphic artist for CBS and work on Disney's EPCOT Center, Florida.

"One of the upper executives at Universal suggested that we do a Conan show," Alexander recalls. "So I came up with the basic idea of doing a big special effects show — something that would demonstrate what movie special effects and a movie set look like. People come here and go on the [Studios] Tour, and they will see some production companies working on films. But the [film crew] is almost always standing around doing nothing. That's the way movie production is — you have 85 people standing around waiting for one guy to get the light right. So what we did was put together a 20-minute show, complete with a little story line, that takes movie special effects and does them on a sound stage right before your eyes.

"The first thing I did," he continues, "was watch a whole bunch of special effects movies to see what other people were doing: what looked good and what we might be able to duplicate live. We picked effects we knew would come off well. I knew lasers look great live, as does flame, smoke, water . . . We started from those basic elements and assembled a dramatic plot around them.

"Sound effects were *very* important too. One of the things I don't like about stage plays is that they usually don't have any sound effects. You watch a movie, on the other hand, and it's got all that tremendous sound. We wanted to make sure that was a part of our show too."

Working with directors/script writers Tony Christopher and Gary Goddard, along with an extensive team of theatrical and technical experts, Alexander assembled the show in just three months. As with any large production put together in so short a space of time, there were numerous technical problems to be ironed out. This was especially true in the case of *Conan*, where different teams of experts for each special effect were working side by side. The show's impressive laser effects were designed by Dr. Sandor Holly, and the pyrotechnics by Gene Evans of Astro Pyrotechnics, in conjunction with Don Sweet. The labyrinthine carbon-dioxide gas system behind *Conan's* smoke effects was designed by John Rogers of Aisco and "Ashley"; the show's fire-breathing dragon came from Animated Playhouse in Sylmar, California.

Naturally, coordinating each of these complex facets of the show was high on the list of Alexander's concerns during the set up. "One of the real problems was equalizing and balancing the room, and getting the sound system to work with all the other things we have in the show," he remembers. "For example, we used a Xenon projector lamp for one lighting effect. We found that the leads from the lamp were projecting RF energy into the house, and it was feeding back into the sound system; it just blew everything away. Those are the sorts of problems we had to deal with constantly."

Pre-Recorded Sound Sources

The musical score for Universal's live *Conan* show was written by Basil Palidorius, who also scored the film of the same title. Music for the Universal presentation was recorded and partially mixed in London, a phase of the project that presented some complications of its own, according to Jerry Laidman of Sound Chamber, the show's audio consultant.

"The formatting of the recording was quite complicated," Laidman says, "because it was being done in such a

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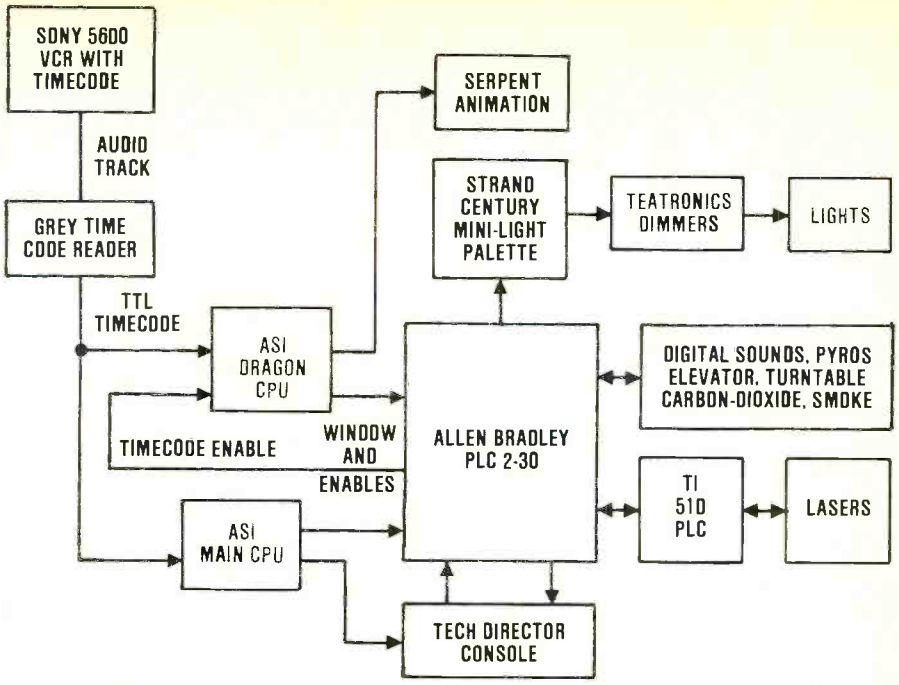
track plays a vital role in the show by providing the controlling sync pulses to an extensive system of computers that govern the timing of the show's audio and visual effects.

Digitized Special Effects

This master digital tape, however, is only one of several pre-recorded sound sources for the show. Some 32 sound effects are digitally stored in bubble memory in an Anitech Systems computer, which serves as the central component in the show's computer-controlled system. The idea of storing sound effects in bubble memory was developed as a means of coping with the split-second requirements of coordinating visual effects — laser zaps, explosions, etc. — with their accompanying sound effects. Each visual effect in the show can occur anywhere within a given time window of several seconds, depending on the timing of the live action occurring on stage. At the precise second a visual effect occurs, the accompanying sound effect has to fire instantaneously. Conventional tape and cart machines were ruled out as storage media for replay of such effects because of the time delay involved in engaging motors and playback heads, rolling past leaders, etc. Bubble memory presented a convenient solution to the problem.

"The bubble circuitry itself was designed by INTEL, and then Anitech made all of its own boards, all the software, and all the card cages," explains Anitech president Steve McIntyre. "Each bubble has a total of 25.5 seconds of audio on it and you can have up to eight channels on each bubble. You can have one channel for the whole 25.5 seconds, for example, or two channels for 12.5 seconds each, or whatever; you can divide it into any kind of combination you want over eight different channels [at 0.1 second resolution]. There's a directory structure that enables you to determine which tracks are played back when. While any one track [within a bubble CPU board] is being played back, none of the others can play back."



COMPUTER-CONTROL SYSTEM SCHEMATIC

The *Conan* installation required five bubble memory boards for audio effects, plus an additional two boards with animation data for the dragon. Such a configuration provides a total capacity of 40 digital effects "tracks," only five of which can be replayed simultaneously. The sound effects, which had been recorded originally in analog, were transferred to bubble memory at the job site by replaying the master PCM videotape, and redigitizing the analog tracks into bubble memory. According to McIntyre, the effects were encoded into bubble memory at a 10 kHz sampling rate, which makes for a playback frequency range of approximately 5 kHz.

Rounding out the show's battery of pre-recorded sound sources are two Broadcast Electronics Spotmaster NAB cart machines, which contain three, longer sound effects associated with the laser. Because these effects exceeded the bubble memory's 25.5-second capacity, it was necessary — and practicable — to put them on cartridge.

Both the soundtrack music and effects underwent fairly extensive revisions during the early stages of the show, audio consultant Jerry Laidman recalls. "Because the show was put together unbelievably fast, there was *no* rehearsal time for anything, and that meant changes later on. For example, it turned out that the dragon's head didn't move as fast as they thought it was going to be able to. We had based all of our scripting on what they had told us. When that didn't work out, we had to improvise and put part of the dragon's sounds on bubble, so that it would be ready to fire at the right time. There was a lot of pulling sounds out, adding them in other places, making them shorter, and things like that."

Script changes also necessitated editing the score, according to Laidman, but Sound Chamber was able to make both the music and effects edits without altering the SMPTE timecode numbers that provide synchronization for the entire show. "We were always able to keep the SMPTE track the same in reference to the music," he explains. "What we had to do was insert different parts of the music in different points, and delete other passages."

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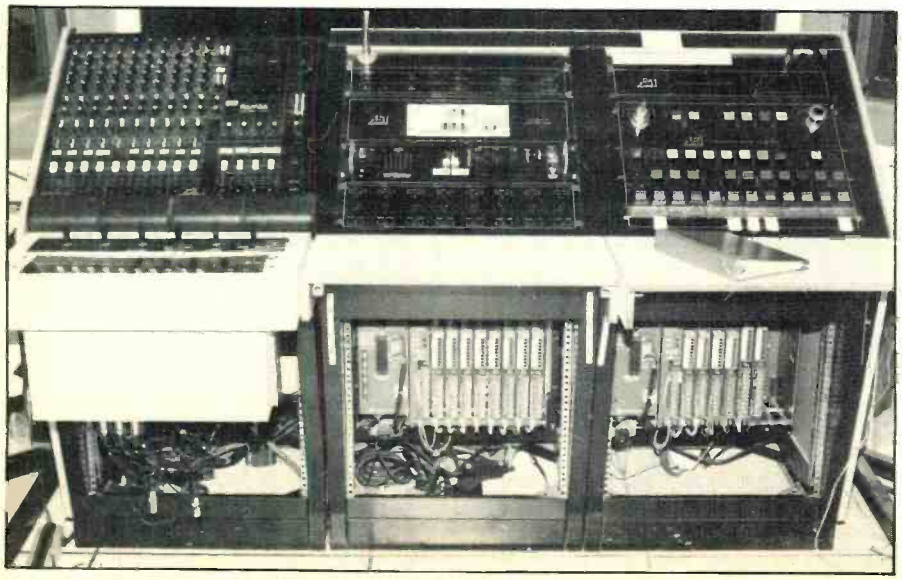
Control area includes a RAMSA WR-8219A console (left), & elements of the Anitech Systems computer, including operator's panel for sequencing & 'arming' special effects.



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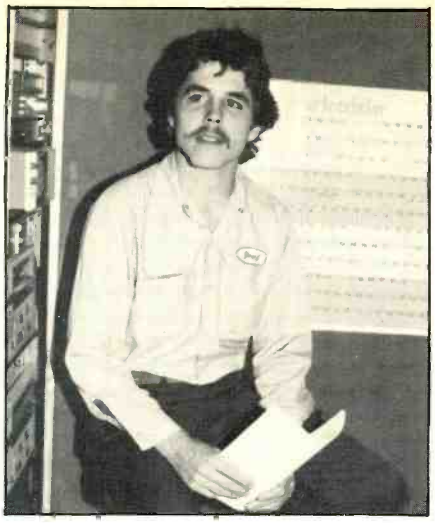
Computer Control System

Virtually every aspect of the *Conan* show is controlled by a system of five discrete computers, which work in concert to coordinate audio and visual effects, and ensure that potentially hazardous laser and pyrotechnic effects are fully safe for both actors and audience. SMPTE timecode from the soundtrack videotape is read by a Gray Laboratories timecode reader, and fed into Anitech's main show computer, which was custom-designed for the show based on the IEEE-796 computer bus standard. The Anitech provides control impulses to several other computers responsible for various show functions. Among these is an Allen Bradley PLC 2/30 programmable logic controller.

"The Anitech follows the SMPTE explicitly," explains John McAdams, who worked with Anitech in developing the show's computer system, pre-programmed all the Allen-Bradley equipment, and now heads up Anidroid Consultants, which specializes in theatrical computer-controlled systems. "It then puts out little 'windows' of time in which a given effect is to take place. During that time window, the Allen Bradley [checks the sequencing] and makes logical decisions as to whether or not all the requisite conditions exist for the effect to take place, and will either fire or not fire the effect accordingly. In other words, the Anitech computer is simply playback. It simply says 'Now is the time,' and that's it; the Allen Bradley is the logical device — it coordinates different effects. If, for example, an explosion does not take place — due to safety constraints, improper arming of the pyro effect, or whatever — then the Allen Bradley will also make sure that the sound effect accompanying the explosion isn't fired either."

In this manner, the Allen Bradley PLC 2/30 controls and coordinates the show's sound effects, pyrotechnics, carbon-dioxide smoke effects, plus the stage elevators and turntables that facilitate the transformation of characters and similar dramatic events. The PLC 2/30, in turn, provides control impulses to a Strand-Century Mini Light Palette, the computer devices that provides automated pre-programmed control of the show's Teatronics lighting dimmer modules.

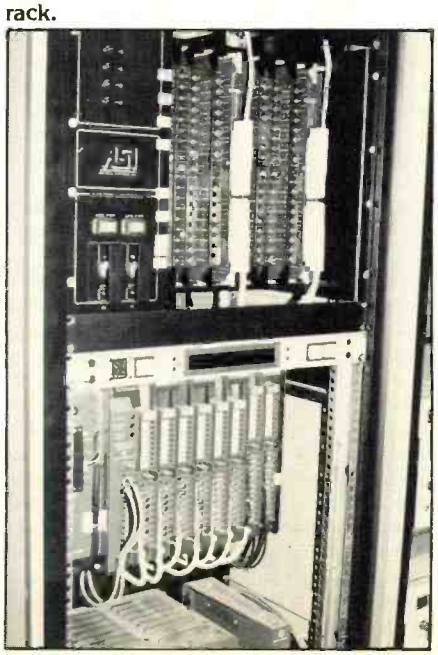
According to McAdams, the decision to have the Allen Bradley computer, rather than the Anitech, provide control impulses to the Strand-Century lighting controller was purely a practical one. "Both the Anitech and the Allen Bradley require that you bring in a programming console, hook it up and program the computer for the lighting cues. But Universal owns the programmer for the Allen Bradley; so if they wanted to 'massage' the timing or something like that, they could play games with the Allen Bradley. It would take a whole team of people, on the other hand, to come in and program the Anitech, because Universal does not own the programmer for that computer."



Yosef Levison, who served as sound/video technician and laser safety officer.

Laser effects for the *Conan* show are governed by yet another computer, a Texas Instruments 510. "The reason why we added the TI for the lasers is because that part of the show was a separate system under a separate contractor," McAdams comments. "It could all have been done on the Allen Bradley, but due to the time constraints in setting up the show and such, we found it better to have the lasers in a stand-alone configuration where we could test that system even if the other systems weren't up and running yet."

To the rear of the sound mixing and control area is the main ASI system computer rack.



The TI 510 receives its timing impulses from the Allen Bradley. This configuration enables the lasers to be coordinated with all the other effects, and also forms part of the show's extensive safety system. The visual effects are generated by two Spectra Physics argon lasers. Rated at 25 watts each, but generally operated at 10- to 12-watt levels, such lasers can seriously injure or blind anyone who gets in their path. Therefore, a thorough safety system was considered mandatory.

"In any given situation where there is an effect involving lasers or pyrotechnics [also a potential source of danger], there are a series of safety features involved in the whole system," explains Yosef Levison, sound/video technician and laser safety officer for the show. "The first thing that happens is the Anitech main show computer will say, 'Okay, we have a window'; that is, a space between two different timecode addresses during which the effect can happen. The time window — say 10 seconds — allows for slop, or for an actor being behind, or a problem on the stage. It allows the actors time to get into position.

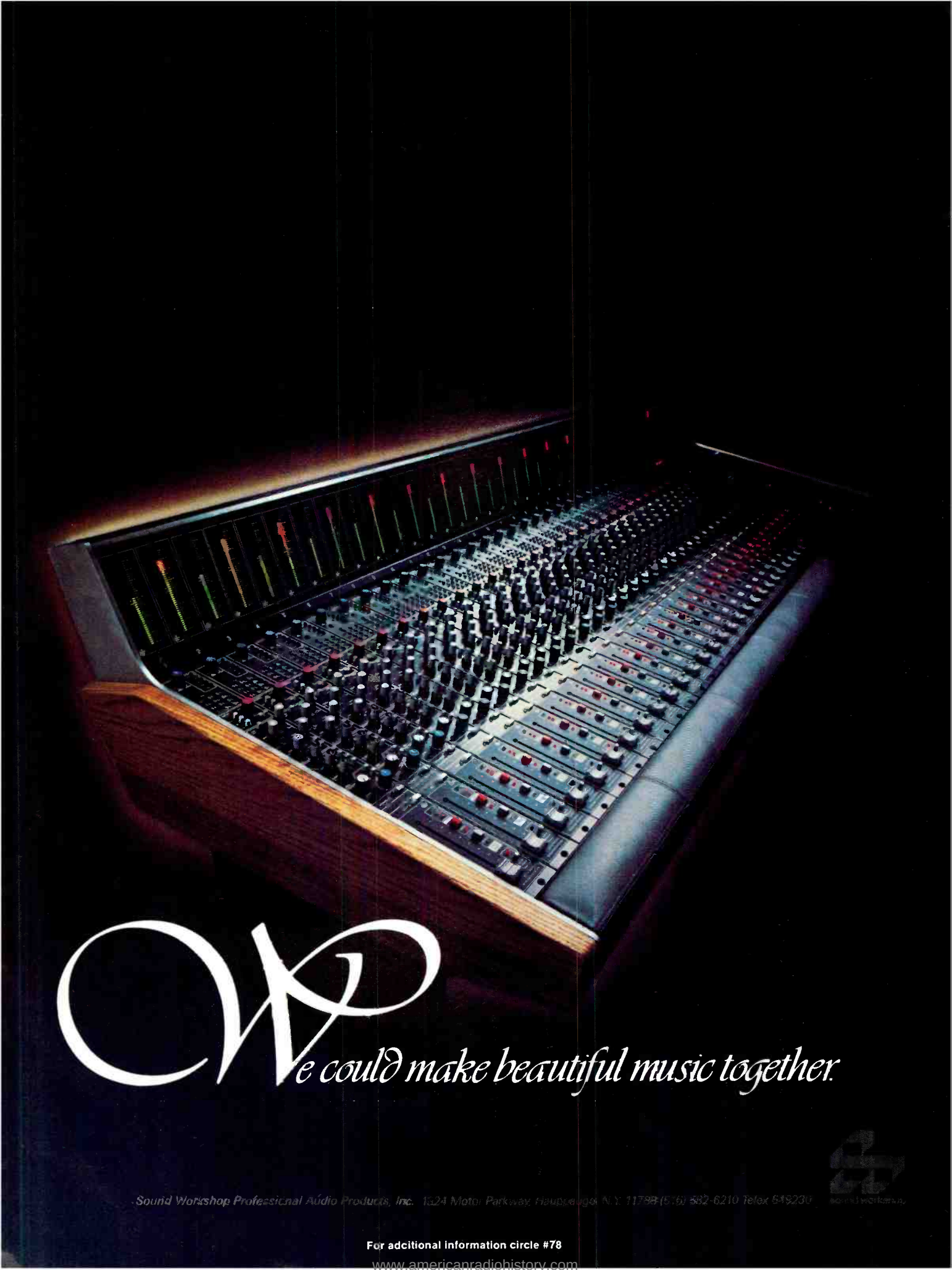
"When that happens, a light goes on at the technical director's control panel, and he will start watching the stage for safety. The actors *must* be standing in specific spots on pressure-sensitive pads which confirm to the Allen Bradley computer and technical director that they are on their mark and therefore in a safe area. We also use ultrasonic and infrared detectors which further confirm that each actor is where he's supposed to be, and that nobody is in any danger.

"When everything looks safe, and all the necessary indicator lights are on, the technical director will press his enable button for that effect. In some cases, this button is the trigger for the effect; in other cases, the actors themselves trigger the effect — either by breaking an invisible beam, or pressing a button."

The show also includes a second custom-designed Anitech computer (also based on the IEEE-796 bus standard) which controls the show's animated dragon. This computer, which also receives safety data from the Allen Bradley PLC 2/30, is enabled by the 2/30 16 minutes into the show via SMPTE timecode, when the dragon is scheduled to make his appearance.

"What we do is manipulate the SMPTE data sent to the second Anitech computer by adding another digit," says John McAdams in explanation of the triggering mechanism for the computer. "We introduce a one-hour bit after 16 minutes, so that the computer thinks it is seeing one hour and 16 minutes [01:16:00]. That one-hour bit is actually a relay controlled by the Allen Bradley. If the PLC 2/30 sends a zero, it prevents the second computer from starting its sequence."

. . .*continued overleaf* —



Here too, it is the PLC 2/30 computer that makes logical decisions, based on input from the show's other systems, as to whether the dragon segment of the show can take place. The dragon emerges from a center stage pit into which several actors fall earlier in the show. There is a stunt mat that must be rolled out of place before the hydraulic lifts that implement the dragon's rise from the pit can be engaged, as John McAdams explains: "The Allen Bradley has to decide, 'Yes, the stunt pad has been rolled out of the way and everything is clear; the air pressure is good and all the conditions prevail that are necessary in order for you to get a complete serpent show.' If any of those things aren't happening, it won't give the second Anitech computer its one-hour bit and you won't have the serpent. You won't have the end of the show either, in that case, but then you *definitely* don't want to bring that serpent up into the stunt pad."

Live-Performance Sound

Although it contains much pre-recorded and pre-programmed material, the *Conan* show remains essentially a live performance. The actors on stage have to be miked and their dialog made audible in every corner of the 2,000-seat theatre in which the extravaganza takes place. The show's vigorous action sequences made wireless microphone systems a natural choice, and what ended up in use was a customized hybrid of wireless equipment.

"We're using HME wireless transmitter/receiver sets with Cetec Vega cases," explains Levison. "We had a problem with the metal cases the HMEs come in, which were deforming because of the falls and stunts. We sent the HMEs out for repairs and, while they were out, we were using the Vegas. We found that their styrene cases were very strong and didn't deform. We knew the HME would fit inside the Vega case, so we changed the connector and hard-wired the antenna — because we had also had some problems with that connector breaking on us. We took out the power switch too, which was a problem with both units. What we wound up doing was simplifying the system so that there were as few breakable components as possible. Since then, we haven't had *any* problems."

The actors wear Sony ECM-150 lavalier microphones. The stage setup also includes a pair of AKG shotgun mikes for picking up various on-stage sounds. An additional mike mounted in the lighting grid is used for an off-stage part that precedes one character's entrance.

"The stage set represents a subterranean chamber, and the character is supposed to be outside [and] above the chamber's entrance," Levison explains. "She's up there speaking into a microphone tied directly to a speaker mounted close to it. It does indeed sound like she's up above the audience."

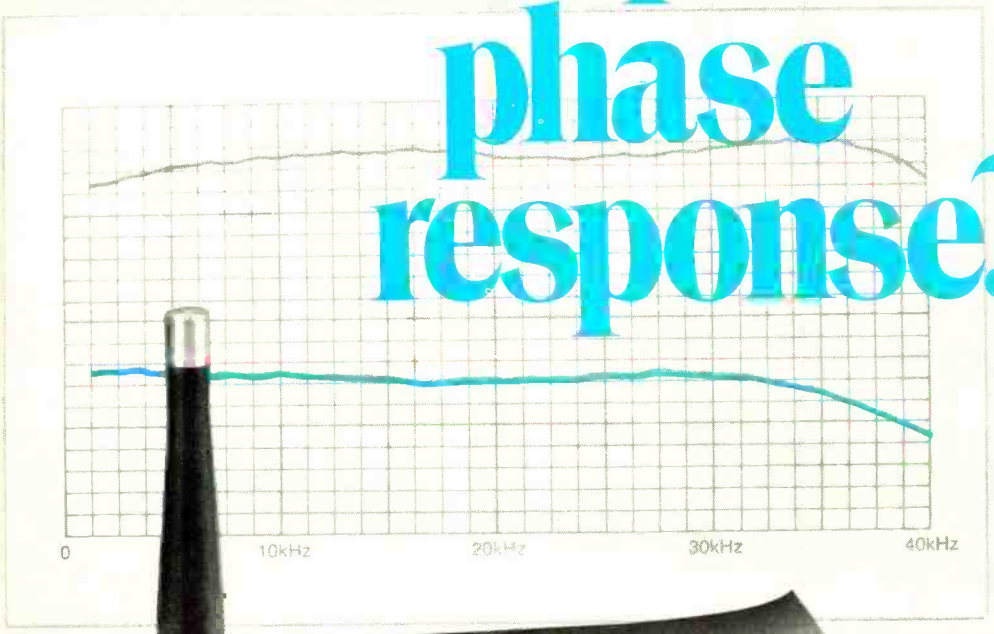
Strand Century Mini-Light Palette unit, which receives lighting control impulses from Allen Bradley PLC 2-30 computer.

All of the live mike inputs are sent to an Industrial Research Products Voice-a-Matic automatic mixer, which allows the crew to pre-set volume, gating and compression threshold levels for each microphone. All mike inputs are summed into a single output, and an Industrial Research Level-Matic module provides levelling for the overall program. The output of the Voice-a-Matic mixer is then processed through a dbx Model 165 compresser-limiter and one channel of an MXR dual 15-band graphic EQ before being brought up on a single channel of the main house mixing console.

Computer control also extends to the show's live miking. The input for each mike can be set in either an on, off or auto mode; in the latter mode, each microphone is only switched on when an actor has a line of dialog to speak. The mike is automatically turned off when the actor is backstage, or engaged in non-speaking action onstage. This function is performed by the Anitech main show computer, which puts out time windows, based on SMPTE time-code addresses, to control relays that open and close each mike channel.

The main house mixing board is a RAMSA Model WR-8219A. Audio for the *Conan* show occupies nine of the console's 10 input channels, plus two effects inputs. The first two channels are devoted to the stereo music/narration track, channel #3 is for the offstage mike located in the lighting grid, and channel #4 is for the Sensurround level (discussed below). Channels 5 through 7 control levels for the bubble memory sound effects, and channel #8 is the dragon's voice, which is replayed from one of the U-Matic's analog tracks. The technical director has a microphone in the sound booth that is used for announcements before and after the

How important is microphone phase response?



The ability of a microphone to reproduce transients without smearing is directly proportional to the uniformity of its phase response. Yet, few manufacturers measure the phase performance of their microphones, and fewer still publish these data.

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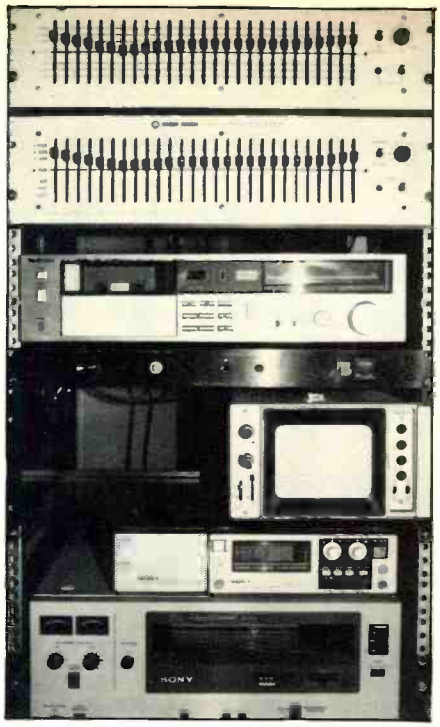
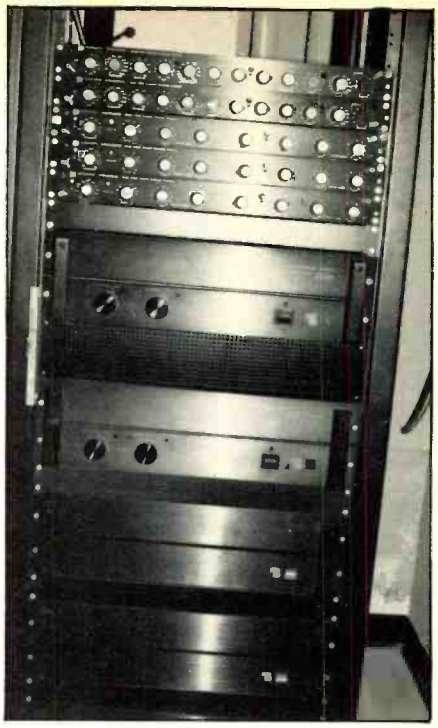
show, and occupies channel #9. Channel #10 is the master fader for the live actors' microphones. Two effects inputs are used to handle the laser sound effects replayed from the cartridge machines.

The Anitech main show computer provides additional mixing assistance by controlling outboard dbx Model 202 voltage-controlled attenuators, which regulate the level on the stereo program channels (#1 and #2) and the Sensurround channel (#4). Along with sound effects, all program information below 70 Hz is fed into the house's Sensurround system via an AudioArts parametric crossover system.

"On the Sensurround and the two main program channels," says Levinson, "we basically just set the zero level, and any trimming is done automatically by the VCAs under computer control."

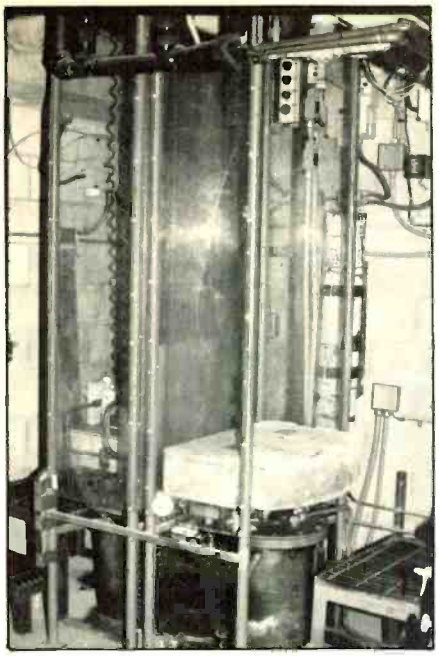
The house PA system consists of six Cerwin-Vega 15-inch bass drivers and six compression horns mounted on the lighting grid above the stage. This rig is supplemented by four Cerwin-Vega 048SES "Earthquake Cabinets" for the Sensurround, mounted in each corner of the audience. The speakers are powered by AB Systems amplifiers, and the house program equalized using a Klark Technik DN27 third-octave graphic equalizer.





Sound equipment racks house Klark-Technik DN27 third-octave graphics, Technics cassette deck, Sony PCM-F1 processor and U-Matic VCR (left), and a bank of Audio Arts crossovers and AB Systems power amplifiers.

In a show where actors coordinate live dialog and action with audio and visual effects under pre-programmed control, accurate and dependable timing is



A dual-action lift mechanism used during the Conan "Transition scene."

essential. Should a time window for a given effect expire before the actors all reach their appointed safety mats, the effect cannot take place. And, since the actors take all of their cues from the music track, on-stage monitoring is naturally a prime concern. Foldback for the actors is provided by a pair of Cerwin-Vegas cabinets mounted in the lighting grid behind the house speakers, and pointing down at the stage. The foldback program is equalized using one channel of an MXR dual 15-band graphic EQ.

"We're running the foldback as a mono submix right now," says Levison, "but we're thinking of making it stereo. The actors have to respond to the music cues, and they have to be able to respond to each other. So anything we can do to improve the monitoring is certainly worthwhile. This is such a large, echo-laden house that good monitoring is vital. We've had it happen that the monitor system has failed, and the delay time in the room has thrown the actors."

The Show Goes On

As the *Conan the Barbarian* show enters its second year, the frenzied activity of its installation and initial months have settled into a daily routine of maintenance, calibration and system improvements. The entire show tape is run every morning in order to reset the computers for their day's work, and provides an opportunity for the crew to check the entire system. There is also a daily safety procedure that the Federal government requires for the lasers and pyrotechnics. The shows begin at 12:15 p.m. and runs at regular intervals until 6:25. Before each show, the actors assemble onstage for a microphone check.

The crew has taken advantage of the more relaxed pace that prevailed in recent months to improve and upgrade the sound system. Installation was recently completed, for example, on a second, backup U-Matic videocassette machine.

"What we're planning on doing is running the second videotape 5 or 10 seconds behind the main program," explains Levison. "That way, if one machine fails, we can just switch over to the other system. The SMPTE reader will freeze as soon as the other machine comes on because the timecode number on the backup tape will be behind the main tape. It will continue counting as soon as the tape catches up 5 or 10 seconds later, and the show will continue from there."

Apart from such enhancements, the show's systems are undergoing a continual process of streamlining. "We're in the process of improving the last few things," Levison concludes. "In the sound system, we've rewired and reassigned a lot of the sound. Now that we have a handle on exactly what we want to do, we've taken out equipment that was installed but which we found unnecessary — things like extra equalizers and summing amplifiers. We're rewiring things that should have been put in with a little more care, but were set up in such a hurry that there just wasn't time then. We've got everything just about where we want it, so we're very comfortable right now." ■■■

CONSTRUCTING PADS AND ATTENUATOR CIRCUITS

by ETHAN WINER
By-Word Corporation
E. Norwalk, Connecticut

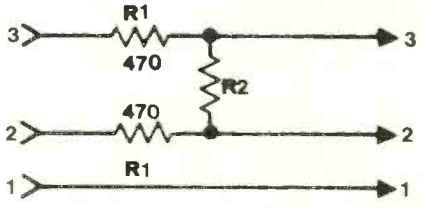
In past issues of *R-e/p* we've looked at a wide variety of construction projects, some of which have been fairly complex. But, as likely as not, it will be the more simple problems that a studio maintenance engineer must contend with on a day to day basis. With that in mind, over the past year or so I have been compiling an assortment of little projects, most of which can be built in an hour or two. It is these "miniature marvels" that are the subject of this month's article.

One of the things that first got me interested in building these diminutive devices is the S3FM connector assembly made by Switchcraft. A simple device really, the S3FM is a 3.5-inch tube fitted with a male XLR at one end, and a female at the other, with enough room in the middle to build a mike pad, a low-end rolloff network, a polarity reverser, or any of a number of useful circuits. Related to the S3FM is the 380AP1 Series of XLR-to-¼-inch audio adapters. The metal sleeves on these adapters are quite a bit smaller than the S3FM, although I did manage to squeeze a phantom-powered active direct box into one of them. Placed at the end of a standard mike cord, you can plug it directly into a bass or synthesizer, without the usual multitude of cables, adapters and clutter.

Of course, there is no reason to limit ourselves only to gadgets that can be shoe-horned into a miniature connector. So, along with the pads, rolloffs, and the direct box, we also will look at a battery-powered phantom supply, and a mike/line level test oscillator. That last circuit will come in handy when you're by yourself setting up for a remote session,

. . . continued overleaf —

FIG. 1: PAD



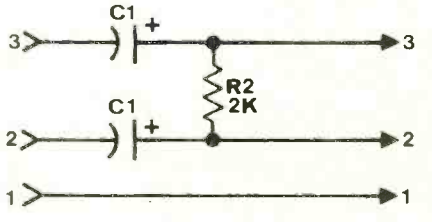
$$R2 = \frac{R_{IN} \times R}{R_{IN} - R}$$

$$\text{WHERE: } R = \frac{2\,R1}{(10^{A/20}) - 1}$$

A = DESIRED LOSS IN DB

R_{IN} = YOUR CONSOLE'S INPUT IMPEDANCE

FIG. 2: L.F. ROLLOFF

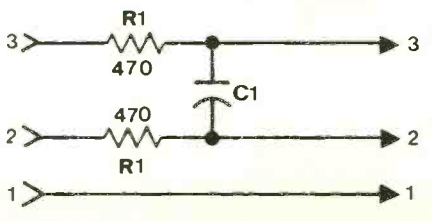


$$C1 = 2 \times \frac{1}{2\,\pi\,FR}$$

WHERE: F = DESIRED CUTOFF FREQ.

$$R = \frac{R_{IN} \times 2000}{R_{IN} + 2000}$$

FIG. 3: H.F. ROLLOFF



$$C1 = \frac{1}{2\,\pi\,FR}$$

WHERE: F = DESIRED CUTOFF FREQ.

$$R = \frac{940 \times R_{IN}}{940 + R_{IN}}$$

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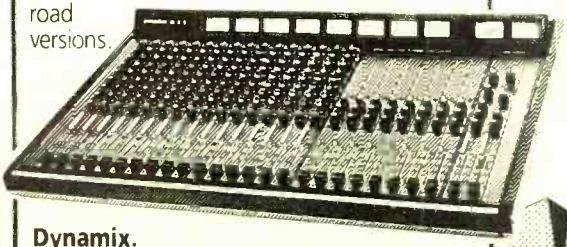

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and need to verify that the mike cables and console or recorder inputs are functioning properly. But let's start at the beginning with the easier items first.

Pads and Rolloff Circuits

The balanced mike pad shown in Figure 1 is about as basic a circuit as you are likely to encounter as an audio engineer. However, as we saw in the February issue, while discussing op-amp gain and loss networks, finding the optimum resistor values can be a challenge for the beginner, and a time consuming nuisance at best for the more advanced. So once again, the computer is a handy friend to have around, and in the accompanying sidebar is given what I refer to as my "Universal Audio Network Program," which will handle the calculations associated with the various pad and rolloff circuits described in this article.

Figure 2 shows the same general configuration, except that this time the circuit provides a low-end rolloff beginning at a frequency dictated by the resistor and capacitor values. If you're real sharp, you'll notice that the designation of "R1" does not appear in this diagram. No, the layout artist didn't goof; the same numbering scheme is used throughout all of the figures to make the formulas easier to understand, and also for overall consistency.

FIG. 4: MIDRANGE CUT

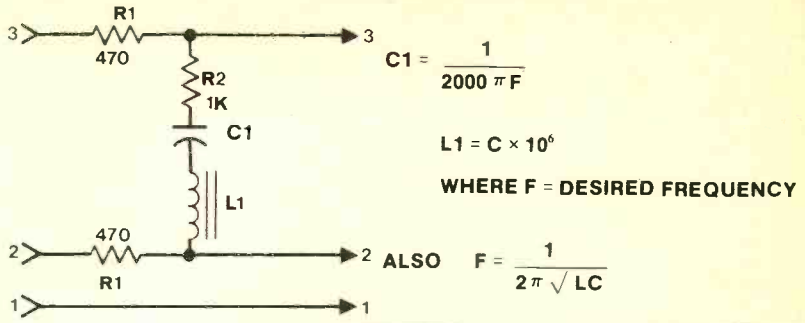
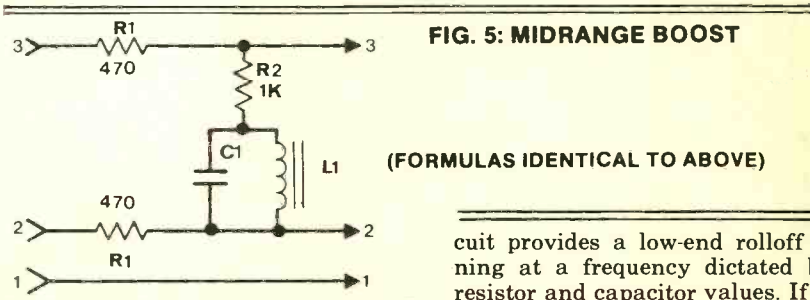


FIG. 5: MIDRANGE BOOST



Transposing the capacitors and resistor gives us a high-frequency loss (Figure 3), while the addition of an inductor allows tampering with the midrange response for boosting "presence," or controlling sibilance (Figures 4 and 5). I should mention that a good source for miniature audio inductors is Mouser Electronics (11433 Woodside Avenue, Santee, CA 92071). In fact, unless your studio has a hand-winding facility, I have yet to discover another source for the type of inductor needed for these circuits. Sure, electronic supply houses carry all kinds of coils, but just try to find one larger than 1 milliHenry or so, especially for less than $10.00. And then there's the other problem of dealing with the inductor's inherent series resistance, which is inevitable due to the wire with which they're made.

The smaller (physically) a given inductor is constructed, the higher the resistance will be, since the wire must be a smaller diameter. The Mouser 43LH and 43LJ Series of inductors effectively circumvent this limitation by winding the wire around a ferrite core. For high-level processing — as in a speaker crossover — iron or ferrite cores generally are to be avoided, since the core material can overload or "saturate," just like magnetic recording tape. But at these extremely low currents the ferrous core does a great job of increasing the available inductance without requiring as many turns, which keeps the series resistance relatively low. Also, these coils are encapsulated into a rugged phenolic case, which eliminates the danger of damaging the hair-thin wires during assembly.

For the midrange networks shown in

BASIC PROGRAM TO CALCULATE FILTER AND ATTENUATOR COMPONENT VALUES

If you follow through the formulas used in the BASIC program listed below, you will notice a departure from the traditional approach of designing this type of network. In the early days of audio, a great deal of effort was made to ensure that all source and terminating impedances were properly "matched." Which means that if a given microphone has an output impedance of, say, 200 ohms, then the pre-amplifier circuit to which it was connected should have an input impedance of the same value. In late years, engineers

```
]LIST

1  SPEED= 190
2  HOME : VTAB 7: FOR H = 1 TO 40: PRINT "*";: NEXT
3  FOR V = 8 TO 20: PRINT "*": NEXT
5  FOR H = 1 TO 40: PRINT "*";: NEXT
6  FOR V = 8 TO 20: HTAB 40: VTAB V: PRINT "*": NEXT
8  VTAB 3: HTAB 5: INVERSE : PRINT "ETHAN'S UNIVERSAL NETWORK PROGRAM": NORMAL
10  VTAB 10: HTAB 4: PRINT "SELECT FROM THE FOLLOWING CHOICES:"
12  VTAB 13: HTAB 9: PRINT "1>  MICROPHONE PAD"
14  HTAB 9: PRINT "2>  LOW END ROLLOFF"
16  HTAB 9: PRINT "3>  HIGH END ROLLOFF"
18  HTAB 9: PRINT "4>  MIDRANGE REDUCTION"
20  HTAB 9: PRINT "5>  MIDRANGE ENHANCEMENT"
21  HTAB 9: PRINT "6>  QUIT"
22  SPEED= 255:PP =  PEEK ( - 16368): VTAB 10: HTAB 38: GET A$:A =  VAL (A$)
26  ON A GOTO 1000,2000,3000,4000,5000,6000: GOTO 22
1000  ONERR  GOTO 1000
1100  HOME : VTAB 5: PRINT "ENTER THE DESIRED ATTENUATION IN DB: ";: INPUT "";A
1110  GOSUB 10000
1160 A = 10 ^ (A / 20)
1200 R = 940 / (A - 1):R2 = (R * RIN) / (RIN - R):R2 =  INT (R2 + .5)
1300  PRINT "R1 = 470 OHMS": PRINT "R2 = ";R2;" OHMS"
1400  VTAB 23: HTAB 5: PRINT "<PRESS ANY KEY TO RUN AGAIN> ";: GET Q$: GOTO 2
2000  ONERR  GOTO 2000
2010  HOME : VTAB 5: PRINT "ENTER THE LOWER CUTOFF FREQUENCY: ";: INPUT "";F
2020  GOSUB 10000
2060 R = (2000 * RIN) / (2000 + RIN):C = 2 * (1 / (6.28 * F * R))
2070 C = C * 10 ^ 6:C =  INT ((C * 10) + .5) / 10
2075  PRINT "C1 = ";C;" MICROFARADS": PRINT "R2 = 1000 OHMS"
2100  GOTO 1400
3000  ONERR  GOTO 3000
3010  HOME : VTAB 5: PRINT "ENTER THE UPPER CUTOFF FREQUENCY: ";: INPUT "";F
3020  GOSUB 10000
3030 R = (940 * RIN) / (940 + RIN):C = 1 / (6.28 * F * R)
3040  GOSUB 9000
3050  PRINT "R1 = 470 OHMS": PRINT "C1 = ";C;C$
3060  GOTO 1400
4000  ONERR  GOTO 4000
4010  HOME : VTAB 5: PRINT "ENTER THE FREQUENCY TO ATTENUATE: ";: INPUT "";F
4020  POKE 216,0
4030 C = 1 / (6.28 * F * 1000):L = C * 10 ^ 6:L =  INT (L * 1000 + .5)
4040  GOSUB 9000
4070  VTAB 15: PRINT "C1 = ";C;C$: PRINT "L1 = ";L;" MILLIHENRIES"
4080  PRINT "R1 = 470 OHMS": PRINT "R2 = 1000 OHMS"
4090  GOTO 1400
5000  ONERR  GOTO 5000
5010  HOME : VTAB 5: PRINT "ENTER THE FREQUENCY TO BOOST: ";: INPUT "";F: GOTO 4020
6000  HOME : VTAB 15: HTAB 3: PRINT "TYPE <RUN> TO RE-ENTER PROGRAM": END
9000 C = C * 10 ^ 6:C$ = " MICROFARADS"
9010  IF C < .01 THEN C = C * 10 ^ 6:C$ = " PICOFARADS":C =  INT (C + .5): GOTO 9030
9020 C =  INT ((C * 1000) + .5) / 1000
9030  RETURN
10000  POKE 216,0: VTAB 10: PRINT "ENTER YOUR CONSOLE'S INPUT IMPEDANCE: "
10010  VTAB 12: PRINT "(OR PRESS RETURN IF IT IS 1000 OHMS) "
10020  VTAB 15: HTAB 14: INPUT "";I$: IF I$ = "" THEN I$ = "1000"
10030  CALL  - 998: HTAB 14: PRINT I$;" OHMS": PRINT : PRINT
10040 RIN =  VAL (I$): RETURN
```

Marco Polo uses SONEX a little differently...

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Just listen—1. He uses it in Ecuador. And his is the first station in Quito to import SONEX. "It is perfect. We are promoting it to our visitors and they are impressed. We are satisfied because of the acoustics. But it's also good-looking and easy to put up." Marco Polo Torres Vasco is General Manager at HCM-1 Radio, Quito, Ecuador.

2. "We use SONEX around the drums when recording basic tracks, and around the vocalist during overdubs. It works very well because sound waves are confined, not bouncing off walls." Rick Bacus of The Janet Jameson Band, Linwood, Kansas.

3. "SONEX is impressive-looking, and even better than that, it does what it is supposed to do for acoustics" in the church recording booth. Jim Moore, Audio Department Director, Northland Cathedral, Kansas City, Missouri.

4. Multi-Images Resources in Dallas uses SONEX for soundproofing one wall of a "cavernous" presentation room. "SONEX is above reproach," says George Minton.

5. Tri-Video Productions, Lake Tahoe, Nevada, is 95% location TV. "SONEX ships well and performs beautifully. It's great—we use SONEX for cleaner location recording and sharper pictures," says J.P. Davis, Producer/Director.

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*SUBMITTED BY FOURTH ROUND OF SONEX PHOTO CONTEST WINNERS.

this article, you can expect to alter the response by about 4 or 5 dB, though this could easily be varied by adjusting R2 up or down in value. With the high-and low-end rolloffs, there will be a cutoff slope of 6 dB per octave beyond the chosen frequency. This is not so easy to alter, since additional capacitors and resistors would be needed that would not only complicate the design, but would create a substantial loss in level overall.

While we're on the subject of high- and low-end rolloffs, I'd like to put in my own two cents about what I consider to be confusing nomenclature. Many equipment manufacturers label a low-end rolloff switch as being "high-pass," and the high-frequency cut switch as "low-pass." Now I realize that this terminology has its roots in traditional engineering protocol, although I believe that this kind of labeling is outright misleading when applied to audio equipment. If a switch affects the low frequencies, then it should be called a "low-something," and when it influences the highs, a "high-something." Doesn't that make more sense? Of course it does!

Closely related to mike attenuators are those pads that allow you to patch a guitar amplifier directly into your console. Most guitar players that I know would sooner cut off their left hand than use a direct box, but this provides a means for retaining most of the amp sound, while eliminating any chance of sound leakage. For this application, just use the circuit from Figure 1 — balanced input and all — although you will need to alter the resistor values somewhat to accommodate the higher input voltage. If you use 10K for R1, and a 100-ohm pot for R2, you will be able to accommodate any input level up to 100V before the resistors begin complaining. One important note: check the amplifier's grounding *before* you connect it to your board, or at least turn the monitors way down.

Photo detail of attenuator circuit built into a pin base of S3FM connector shell.

Construction Tips

With space at a premium, you certainly don't want to use resistors larger than the standard ¼-watt types. In fact, for any mike-level application there will be no problem with ⅛-watt resistors if you can find them. I always prefer carbon film resistors over standard carbon for audio work, but they're even harder to find in an ⅛-watt package. And if you really want to get fussy you can use 1% tolerance parts, although it probably won't make a meaningful improvement.

Another space saving advantage with mike-level circuits is that you can use low-voltage capacitors. When the component values are large, as in the low-frequency rolloff circuit, tantalums will be the only choice, and if you look for them you may be able to find some ultra-miniature 6V or even 3V units. Tantalums may cost more than standard electrolytics, although they are not only smaller but also generally of a higher quality. One note: be sure to have the plus terminals "back-to-back" as shown in the schematic, since the caps will be passing AC. This technique will work in any application where polarized caps have to operate either way, and placing the negative terminals together would work just as well. This circuit will *not* pass phantom power, however.

Unless you use shrink fit sleeving over all of the components (not a bad idea anyway), you will need to ensure somehow that the components don't short out to one another. The S3FM comes with a plastic-coated miniature cardboard tube that lines the inside of the case, so you won't have to worry about that. I generally prefer clear shrink tubing because it allows you to see what parts are inside; a definite advantage three years from now when you have to take it apart for repair. Also, avoid using gobs of black tape for insulation — not only can't you see through it, but after a while it turns into a gooey, sticky mess that is difficult to remove. In fact, the best solution yet would be to draw the schematic on a small piece of paper, and roll that up as the insulation instead.

If you really want to make the assembly secure, and you don't care if it can't ever be opened, you may want to consider encapsulating the whole device in potting compound. Two-part epoxy mixture can be purchased at nearly any electronic parts distributor, although there are several precautions you should follow. First off, it you don't mix the two parts *exactly* right, it may never dry completely. Second, it is possible that some of the liquid will run out the bottom while you're pouring it into the top; a liberal coating of Vaseline around the base of the connector should eliminate that. Instead of standard potting compound, you may choose to use RTV instead. Also available from electronics stores, RTV is a clear, uncured rubber product that dries in about 24 hours. While it never actually becomes hard like the epoxy compounds, it does become quite firm. In fact, that will probably be an advantage since it provides some degree of shock isolation. Also, RTV won't run when you apply it, minimizing the risk to the connector contacts.

FIG. 6A: PHANTOM POWER, SINGLE ENDED

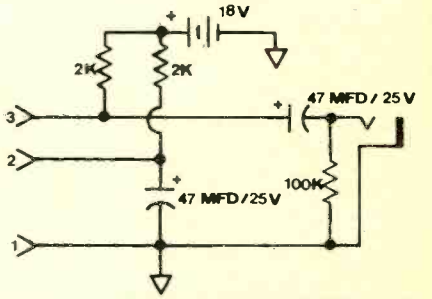
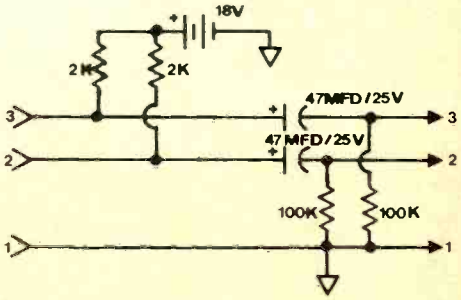


FIG. 6B: PHANTOM POWER, BALANCED



BASIC CALCULATION PROGRAM — continued . . .

discovered that while this was still the best way to go with RF circuits and antennas, a substantial improvement in signal-to-noise ratio could be achieved by terminating the mike into an impedance much higher than its own. (These points were discussed in depth by Paul Buff in his article describing the design and development of the Valley People Transamp mike pre-amp, in the December 1977 issue of *R-e/p.*)

Other benefits can be attributed to a higher terminating impedance, such as reduced distortion and increased frequency response (in some cases), so it was an easy decision to design these pads and rolloffs for a minimum input of 1,000 ohms. Fortunately, this has the additional benefit of greatly simplifying the BASIC program, since the number of variables can be reduced to but a few.

You also will notice that unlike the program shown in the February issue — where the optimum selection of resistors is calculated for you — no such table of standard values exists, and you'll have to pick the closest ones yourself. Again, this is not a serious limitation since we're really going for something quite different this time, and it hardly seems worth doubling the complexity of the program for such a minor convenience. Since you will have to determine the nearest available value, there will be an additional factor when building the midrange networks. If you picked the next smaller inductor value, then you should choose the next *larger* capacitor value to keep the frequency about the same.

Written in Applesoft BASIC, there should be little difficulty in translating this program to TRS-80 or other dialects of BASIC, with the biggest differences being in the print statements (HTAB and VTAB), and the various PEEKS, POKES, and CALLS. The latter can be simply omitted, with no change in the accuracy of the calculations. ■■■

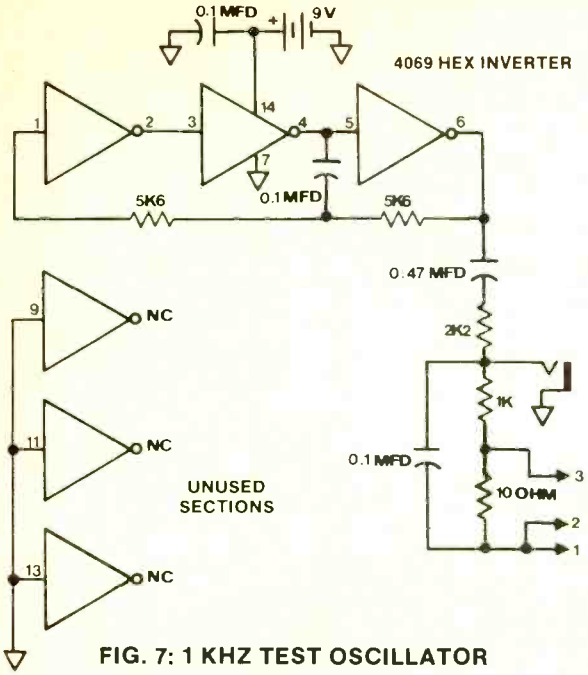


FIG. 7: 1 KHZ TEST OSCILLATOR

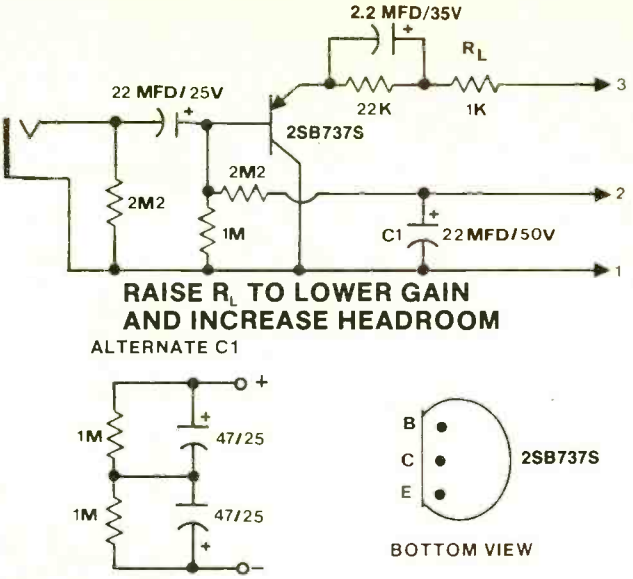


FIG. 8: PHANTOM POWERED ACTIVE DIRECT BOX

FURTHER PROJECTS

Phantom Power Supply

Moving right along, another simple but useful goody is the portable, battery-powered phantom supply for condenser microphones, as shown in Figure 6. Undoubtedly, this will have its biggest appeal for those who do remote recording of, for example, concerts and school plays using a pair of mikes plugged directly into a tape machine. Many AKG mikes can be run on as little as 15V, which allows the use of a pair of standard 9V "transistor" batteries, although most other brands of microphone require the full 48V that is standard throughout the industry.

For mikes requiring a 48V phantom supply, you'll need to use a pair of miniature 22.5V camera batteries which can be purchased at Radio Shack, among other places. This is the same type of battery used by Neumann in their U-87 microphones, and while they won't last as long as the 9V types, they will still provide enough juice to get you through several concerts. It is worth pointing out that when powering an AKG mike from less than 48V, the resistors that connect the batteries to the mike line will have to be adjusted in value. Normally, a pair of 6.8K resistors are used for this purpose, although according to AKG, when using 18V (two 9V batteries in series) these resistors should each be around 2K.

If you do build the 48V version, besides using 6.8K resistors, you will also need to increase the voltage rating of the 47 microFarad electrolytic caps. A 50V rating would probably be alright, although for the highest reliability I would recommend getting caps good to 63V, or more. Also, if you are planning on powering two mikes at a time, you should use two separate battery assemblies. Not only will this extend battery life, it also will minimize any chance of crosstalk. If you insist on using one set of batteries for both mikes, at least place a 47 mFd capacitor between the most positive terminal and ground, which will maintain a low impedance at the point where the mikes could couple together. With fresh batteries it is unlikely that there would be a crosstalk problem, but as a battery ages its output impedance rises.

Tone Generator

Remember the last time you had to check out a bunch of mike cables with an Ohmeter? It wasn't much fun, was it. Well, the job can be made a lot easier if you have some sort of a tone generator that can be plugged into one end of the cord. If sound is produced at the other end, chances are that the cord is okay, although this kind of test won't tell you if the polarity is reversed. Another use for a tester such as this would be preliminary troubleshooting of a console via the patch panel. If nothing seems to be getting through channel #12, for example, insert the line-level output of the tester into each of the appropriate patch points starting with the last. Work towards the input, and when the 1 kHz tone disappears, you will have determined at which point in the chain the signal is getting lost.

This tone generator circuit is based on a CMOS inverter package, such as the 4069 or the 74C04, and should run for a long time on one 9V battery. Of course, CMOS ICs are mainly intended for use in digital logic circuits, but they are right at home in many analog situations as well. Install two output jacks —one for line-level, and one for mike — as shown in Figure 7, and be sure to wire the XLR connector *exactly* as shown. Even though the test signal is being sent down only one of the two hot leads, the other one is used as an "active" return, and if open will not allow the tone to get through. Of course, this will only be true with a transformer-based mike input.

The 4069 IC contains six individual inverters, but you'll only be needing three. As with any CMOS device, always connect the unused inputs either to ground or plus supply to ensure reliable operation.

Active DI Box

The direct box shown in Figure 8 is based on the 2SB737S transistor, which features extremely low noise and high gain. Although the audio output from this circuit is single-ended, I have never encountered any hum or RF pickup when using it, probably due to its very low output impedance. The circuit actually provides a loss in gain from input to output (not uncommon for a direct box), because a phantom supply simply cannot provide enough current to drive a low impedance mike input at line levels. This would be the case if it were connected directly to a synthesizer, or other high-level output instrument. If your console's mike inputs are based on the Valley People Transamp, or another high impedance transformerless circuit, this would not pose a problem, but many studios do not have that kind of input, and the loss is really of little consequence anyway.

The gain loss may be adjusted by altering the value of RL, with increases in resistance providing a corresponding increase in attenuation. If you do have a high-impedance mike input, then you could either reduce this resistor, or eliminate it altogether. You also could reduce the value of the 22 mFd tantalum caps to 1 mFd or so, although for transformer inputs it's imperative that you don't, or the low-end response will suffer. In fact, if your mike inputs have an input impedance substantially less than 1K, you may actually need to increase them. The only problem you might face is finding capacitors that big

in value, and which also have a high enough voltage rating. In this case you must utilize two 47 mFd capacitors in series, as shown in the schematic, with the 1M resistors used to "force" an equal voltage across each one.

As if the inductors weren't hard enough to find, just try to locate the 2SB737S transistor! It represents the current state of the art in low noise transistor design, and is as hard to find as my Uncle Irving's teeth. But you can get them from Phoenix Systems, Inc. ($2.00 each plus $1.00 shipping and handling for orders under $10.00, from 91 Elm St., Manchester, CT 06040). It's also possible to use other high gain PNP transistors, such as the 2N4403 or the 2N3906, although you should probably try a bunch of them, selecting the one with the lowest audible noise. □□□

MAINTENANCE AND TROUBLESHOOTING

A Mixed Bag Of Maintenance Hints and Tips

In past issues this occasional series of articles has covered a wide variety of troubleshooting topics, mainly dwelling on methods and techniques. In this installment though, I would like to present an assortment of isolated, unclassifiable hints and tips that I have developed (or more likely, stumbled upon) over the years.

Sometimes the most difficult problems to solve are those where you might know that nothing is actually broken, but that something is happening that can't be right either. For example, one day I was helping a friend with a home studio try to figure out what was going on with her console. It sounded fine—no noise, distortion, or anything like that — but she was very concerned because, while one of the heat sinks on the rear panel was warm as usual, the other was cold! Hmmmm . . .

Normally I enjoy a good challenge, but this console was one of those "all in one" units where, in order to gain access to anything, you must remove all of the panels, all of the knobs, and about a million screws. (Guess who got to do that!) Obviously, the console was receiving power, since after all, it was working. (Common mistake number one: *never* assume anything.) So I skipped that and just poked around randomly with a meter. Problem was, I couldn't find anything wrong. Okay, I thought, I'll check the darn power supplies. 117 volts was coming in okay; it got rectified and filtered, and proceeded to the regulators just fine. About 24 volts was being passed to each regulator board — just about what you'd expect for a ±15 volt system.

But then I noticed that on the minus supply, there was the *same* 24 volts coming out! Apparently, the "pass" transistor — the one that actually drops the voltage — was shorted. The board still worked, although a lot of little parts would have been perfectly justified had they gone up in a poof of smoke.

And the lack of heat? Ohm's Law will tell you that in order to dissipate power, you need both amps and volts. Without any voltage being dropped across the power supply transistor, it ran as cool as could be. I congratulated my friend for being so observant, and sent her to the store for the replacement transistor.

While we're on the subject of power supplies, it might be worth pointing out that transistors and ICs aren't the only devices that can short out; electrolytic caps are notorious for such a failure — especially old ones. If you ever plan to overhaul a piece of gear that is more than 10 or 15 years old, expect to change some caps. I remember fixing an old Fender Twin Reverb guitar amp that kept blowing fuses. (Sound familiar?) And the killer with this kind of problem is that you can go through $5.00-worth of fuses in the blink of an eye, and still not get anywhere.

Of course, Fender Twins also are well known for blowing power transformers, so before finding the bad caps, I had disconnected the transformer, the diodes, and nearly everything else. It would be refreshing to someday find something wrong in the first place I looked, rather than the last all the time.

Even aside from catastrophic failure, bad electrolytic caps also are a frequent cause of "sputtering" noises in audio circuits. And they generally go down in value as they age, allowing crosstalk, oscillation, motorboating, and other nasties. In the warm environment of tube equipment, they will age even faster than usual.

Another thing that drives me nuts is the EQ pots on some consoles. In the old days, all consoles had rotary switches to set the amount of EQ boost and cut. While clearly a more expensive way to provide selectable EQ, every module could be guaranteed to be identical, and when the dial showed "zero," you knew it was flat. With many units today, pots are used instead of switches, and you

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For additional information circle #87

are counting on the accuracy of the center detent for a flat setting. Or worse, some sleepy-headed assembly trainee on a Monday morning putting on the knobs. I am not knocking pots, and in fact they have several advantages, but if only there was some easy way to tell if the knob (or detent) is centered.

Trick #2: Apply a medium-level squarewave to the input of the equalizer, while looking at the output with an oscilloscope. Use a 100-Hz or so beat signal, and simply adjust the boost/cut knobs for the straightest lines on the screen, both vertically and horizontally. When the pots are in the middle, you'll know it. Then you could either reposition the knobs on the pot shafts, or make pencil marks on the console panel.

One of my least favorite thing to do in the studio (besides running 250 cassette dupes) is soldering XLR connectors. I'll strip and solder foil-shielded cable all day, but when it comes to braided shield — the only type really suitable for mike cords — forget it! If you can sympathize with this, and would like the wires to last as long as possible, always use the kind of cable made up of many separate tiny strands. Besides being much more flexible than the budget cable available from suppliers like Radio Shack, this wire will last much longer between repairs, and is available from several suppliers around the country.

Now when it comes to soldering XLRs in place, a little care will go a long way. Generally, the weakest place is at the point where the wire is soldered to the terminal. When solder is applied to the heated connection, some of it likes to "wick" up into the strands, especially if the ends have been pre-tinned. If you like, you might want to procure a tool called — you guessed it — an "anti-wicking" tweezer. It resembles one of those little clip-on heat sinks, and in fact serves a similar purpose: in this case, absorbing heat away from the end of the connection, before it can be absorbed by the wire strands. And while you're at it, don't forget to ground the XLR case terminal to pin 1: this guarantees continuous shielding right up to the connector pins.

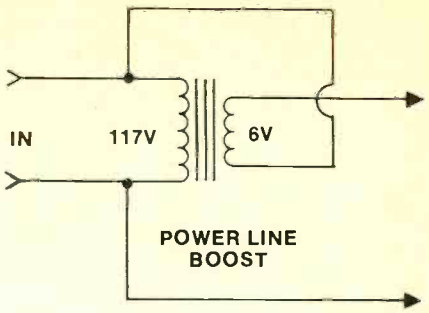


Figure 1A

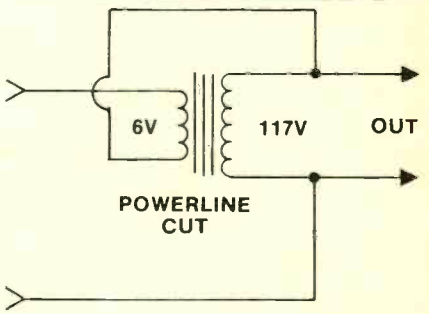


Figure 1B

Speaking of connecting cables, I have seen lots of folks get into serious trouble by using long connecting cables with Tascam or Otari (and other) low-level equipment; it has nothing to do with level of course — generally around -10 dBV — but rather with the medium to high matching impedances involved — typically 10 kohm, and up. Another friend with an eight-track setup in his home asked me over one time to help him track down some missing highs. He had just put the studio together, and couldn't understand why everything sounded so "muddy." I couldn't either, until I saw a huge loop of "extra" cable connecting his console to the tape deck. Since he wasn't sure where he wanted to put everything yet, a hundred feet of cable seemed the perfect solution. That one cost him about 18 dB at 20 kHz! *Read the Manual, Silly!* It's right there in black and white — keep the cables less than 10 feet long when connecting high-impedance inputs and outputs.

Judging by the amount of mail I received when I mentioned power-line filters a few issues back, I'd guess this must be a sore topic for a lot of studio

people. These filters (made by CDE and Corcom) are available from most professional electronic supply houses, and are wonderful for eliminating transients and other noises that ride on the power lines. But that's as far as they go. Before making a big investment, be sure that the cause of the interference is well known, because if it's coming in through the air, you'll need a totally different approach.

Inexpensive SCR-type dimmers certainly send all kinds of junk down the power line, but they also radiate all over the place, with the AC power wiring acting as an antenna. Lose those dimmers fast and replace them with variable transformers instead. You may need to hire an electrician to keep the whole thing legal but, if it gets rid of the buzzing, it'll be well worth the additional expense.

And while we're discussing power lines, there's a great trick for adjusting the voltage up or down slightly — even if you don't have access to the main transformer. I dug this one up years ago when I had a small four-track studio in an old barn out in the country. The place had power alright, if you consider 112 volts adequate. By putting a 6-volt, 10-amp "filament" transformer in series with the line going to the equipment, it was easy to raise the input power to 118 volts; see Figure 1A. It is equally easy to lower the voltage in a similar manner, as shown in Figure 1B. If you can't find a big enough transformer, use several and spread them out: maybe one for the console, one for the outboard rack, and another for the tape machines. Just be careful, and don't get electrocuted. (Or caught, if the zoning laws in your town require a licensed electrician to do the job.)

One of the most difficult troubleshooting assignments is trying to fix a newly built piece of equipment. If you're working on some old amp or tape deck, there is a limit to what might be wrong with the thing. After all, it did work at one time. All that could possibly be wrong would be either components or connections. I mean, what else is there?

Now, on a new piece of home-made gear *anything* could be wrong, including the very concept of the circuit to begin with. (Actually, anything even reasonably complicated should be bread-boarded first; it saves a lot of messy re-work . . . but you get the point.) This was exactly the case with yet another friend who had built a "copy" of a Heathkit amplifier. I had built one from Heath, and it turned out very well, so Phil decided he would just buy the power transformer and a few other key parts from Heath, and do the rest himself. With more available time than money, he did a beautiful job; and it even worked — well, almost.

Everything was great except for this irritating hum on the phono inputs. Even though the basic circuits were a copy of a proven design, the overall layout was designed from scratch, with all of the components mounted to a large piece of Perf board. Of course, we assumed that there must be a ground loop somewhere; these pesky critters can pop up at a moment's notice to plague even the most carefully designed layout.

So we checked ground wires, ground lugs, chassis ground — you name it, we checked it. Anything we tried either made no difference, or made it worse. Until, that is, Phil pushed the turntable aside to get better access to one of the connections at the rear of the amplifier. The hum suddenly became much quieter. He pushed it further and the hum disappeared completely. For crying out loud, the power transformer was putting out a hum field, and the turntable's cartridge was picking it up. There never was anything wrong with the amp in the first place; the turntable was simply too close on the shelf!

This kind of radiated interference can be caused by lots of other kinds of equipment as well — and not just from power transformers. When I first set up my last studio, I had an inexpensive B&K oscilloscope mounted on the edge of the console near some outboard gear. If you listened very carefully, you could sometimes hear a whining background noise that ended up on the final mix. Fortunately, we figured that one out immediately. Digital clocks can do the same thing as well, since there are all kinds of frequencies with fast rise times floating around inside. ■■■

DELTALAB EFFECTRON II ADM-1024 DIGITAL DELAY AND EFFECTS PROCESSOR

Reviewed by Roman Olearczuk

During some recent audio sweetening sessions for the network promotion of an NBC miniseries, *V: The Final Battle*, our post-production sound crew had the opportunity to try out several different audio effects devices. When one of the video editors suggested we use a DeltaLab Effectron®II ADM-1024 processor to create various alien voice effects, there was skepticism that a digital delay unit costing less than $500 could sound good, let alone offer over one second of delay time, a full audio bandwidth of 16 kHz, time modulation and feedback effects, and a dynamic range of 90 dB!

Not only were the above claims found to be true, we also discovered that the device provided all these features packed into a rugged 1¾-inch rack-mount unit. I decided to further evaluate the DeltaLab Effectron II during daily production routines, to test and assess its operational features. This hands-on report discusses the finding of such an assessment, and hopefully will provide the reader with insight regarding the benefit/cost tradeoffs designed into this device.

Input/Output Characteristics

The Effectron II ADM-1024 unfortunately does not provide XLR input/output terminations, although the unity can be easily interfaced to mixing consoles through the use of ¼-inch phone-to-XLR adapters. A "synthesized" stereo output is also available on the ring portion of the ¼-inch jack. According to DeltaLab, the tip of the three-pole jack outputs the delayed signal, while the ring output is via a frequency-dependent, phase-shift circuit. The phase shift begins to take affect above about 300 Hz, and produces a randomized phasing effect described as "very subtle." Panning the two outputs left and right in a mix produces a pseudo-stereo sound, while summed to mono the two signals produce the traditional comb-filter effect.

In addition, there are back-panel jacks for Control, Bypass, and Repeat, which would be useful to on-stage or studio musicians for direct instrument effects, but were not used during our special-effects production.

Output level is not individually adjustable, but is dependent upon the amount of signal level present at the input, as well as the system gain setting on the input-level control. Visual indicators consisting of red and green LEDs allow the user to optimately set the gain through observation. The active (green) LED provides an indication that the input signal is within a 20-dB window of optimum processing. The limit (red) LED alerts the user that the signal is approaching clipping.

Ideally, the active LED should be on continually while the limit LED just begins to flicker. When this condition is achieved, input signal levels can be fully processed over a range of 0.1 to 7.1 VRMS without any sacrifice in dynamic range. Additionally, within these requirements there remains an extra 6 dB of headroom for unexpected peak signals. The output signal is then a nominal 2.0 VRMS at a full dynamic range of 90 dB.

The device also is equipped with a useful Delay Mix feature at its output. The signal mix output can be adjusted from a source-only signal on through an in-phase, delay-only signal by rotating the delay mix control clockwise from a mid-range position. Likewise, out-of-phase delay signal can be added to the source by rotating the delay mix control in the opposite direction.

Delay Time/Effects Description

Delay time is selected by pressing a pushbutton delay increment (1, 4, 16, 64, 256, or 1024 milliseconds), along with adjustment of a rotary knob that functions as a multiplying delay factor valued from 0.25 to 1.0. In this way, any desired delay time between 1 millisecond and 1.024 seconds can be achieved through use of the two interactive controls. Infinite repeat mode is accessed by a separate pushbutton. Even though the unit does not have a time increment display, convenient time-effect labels are provided underneath the delay pushbuttons to aid the user in quick setups. For example, three pushbuttons, designated HI, MID, and LO FLANGE, correspond to the three smallest time delay increments: HI FLANGE covers the delay range from 0.25 to 1 millisecond; MID FLANGE from 1 to 4 milliseconds; and LO FLANGE from 4 to 16 milliseconds. In a similar fashion, the remaining time increments of 64, 256 and 1024 milliseconds correlate with DOUBLE, SHORT ECHO, and LONG ECHO labels.

A feedback control knob allows the user to vary the amount of positive or negative feedback signal to be added to the delayed signal for producing unusual effects. As an example, the intensity and richness of a flange can be increased with maximum feedback, since larger signal peaks and nulls are produced.

Time-delay modulation effects are provided via two controls labeled Width and Speed. The Width control varies an internal oscillator to provide a periodic time delay change, where the sweep range is a percentage of the delay-factor value. In the 100% setting the delay time will vary from just under 0.25-times the delay time to just over 1.0-times the delay time. At maximum width modulation over two octaves of pitch sweep can be obtained via this Doppler-Shift effect. The sweep range is the modulated at a rate from 0.1 to 10.0 Hz according to the Speed control setting. When modulation is not desired, the width control is left in the 0% setting. Both controls can be used to enhance flanging, to provide tremolo effects, and to detune doubled voices.

Operational Comments

The Effectron II ADM-1024 is a high-quality audio product. In fact, in the delay-only mode it was difficult to audibly distinguish the source signal from the processed signal, even at maximum delay times. The unit was quiet during idle times, and it did not add any appreciable noise when patched to console returns. The active/limit LEDS provide a good indication of input signal level, and are accurate at signal clipping levels.

A number of effects were tried with the help of the operations manual, which provides the user with sample effect settings (for example, vibrato, slapback echoes, comb filtering, etc.) through the use of the illustrated front-panel templates. All effects settings provided a professional sound with inaudible signal distortion.

In particular, the "Tuned Resonance" effect produced an unusual metallic electronic speech when a human voice was used as stimulus. Varying the delay factor around the 16-millisecond setting produced different pitch resonance "rings" that were analogous in sound to that produced by bending a metal sheet as it is being struck. The intensity of this effect was increased whenever more feedback was introduced. Slapback short echoes with feedback — both from the device and the console's echo sends — proved to be very effective in attain-

I N T R O D U C I N G

THE TIME MACHINES

from **Garfield Electronics**

The amazing thing about time in a recording studio is how quickly it gets eaten up. Especially when synthesizers, sequencers and drum machines are concerned. That's where The TIME MACHINES from Garfield Electronics come in. They not only control time... they save it!

SYNC TO THE CLICK.

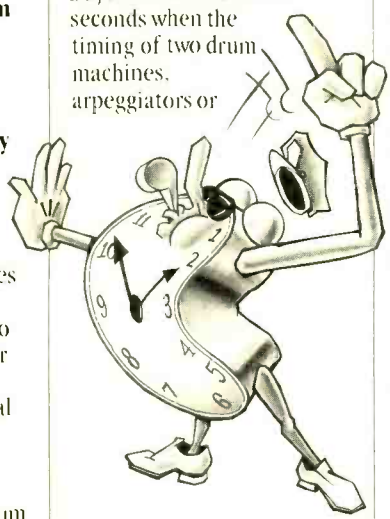
With DOCTOR FLICK, all makes of sequencers and drum machines simultaneously sync to click tracks thru their clock or sync inputs, including Roland, Oberheim, Sequential Circuits, Linn, Korg, Moog, Fairlight, Synclavier, E-mu, Wave PPG, MXR and even MIDI-ized sequencers and drum machines that sync to tape.

Built into The Good Doctor is the most accurate and versatile metronome in the industry, generating click tracks in 24, 25 and 30 frame per second calibrations with 32nd frame resolution for film and video production.

And from audio sources, DOCTOR FLICK generates individual triggers for arpeggiator, gate and individual drum trigger inputs. $1195.00

FINE TUNE THE "FEEL".

The difference between a pushed or laid back "feel" can be just a matter of milliseconds when the timing of two drum machines, arpeggiators or sequencers is even slightly offset.



The high resolution stereo CLOCK DELAY gives you the opportunity to compensate these offsets by fine tuning timing between instruments till you have just the "feel" you're after. $495.00

CONTROL YOUR SHAKE, RATTLE AND ROLL.

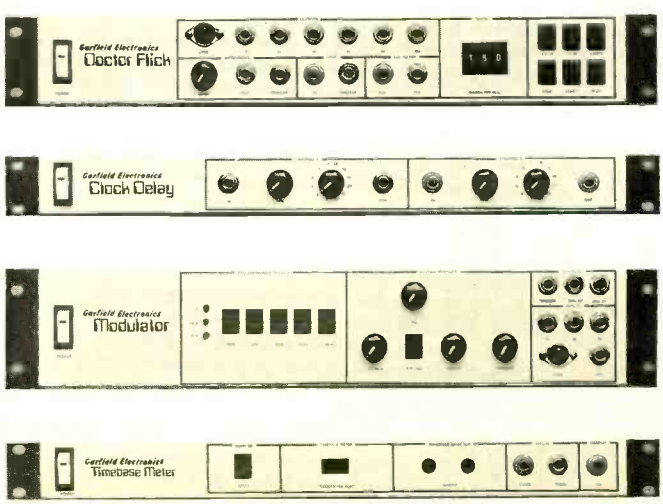
Assume more control over your electronic rhythm section with The MODULATOR original Live Rhythm Sequencer. Programmable Trigger controls arpeggiator, gate and drum machine trigger inputs with auto-corrected rhythm patterns you can create, edit or add to on the fly. And it's Triggered Rhythm Envelope modulates synthesizer VCA, VCF, VCO, Filter, Amplifier and Vibrato inputs in perfect beat with your system. $495.00

CHECK YOUR PULSE.

Metering the various clock rates of your instruments is a job for THE TIMEBASE METER. With it, DOCTOR FLICK can be adjusted to any timebase, and with a CLOCK DELAY, it shows when the timing of two machines is perfectly aligned. It also serves as a pulse counter and tape sync code adapter, and is ideally suited for studio work. $395.00

Armed with the power of The TIME MACHINES, when you're in the studio, time is on your side.

The CLOCK DELAY, MODULATOR and TIMEBASE METER enhance Doctor Flick, Doctor Click Rhythm Controller or Mini Doc Synchronizer based systems. For more information on the complete line of TIME MACHINES, or the name of your nearest dealer, call or write:



Garfield Electronics

P.O. Box 1941, Burbank, California 91507
(818) 840-8939

For additional information circle #90

Our Only Business Is Getting Your Act Together.

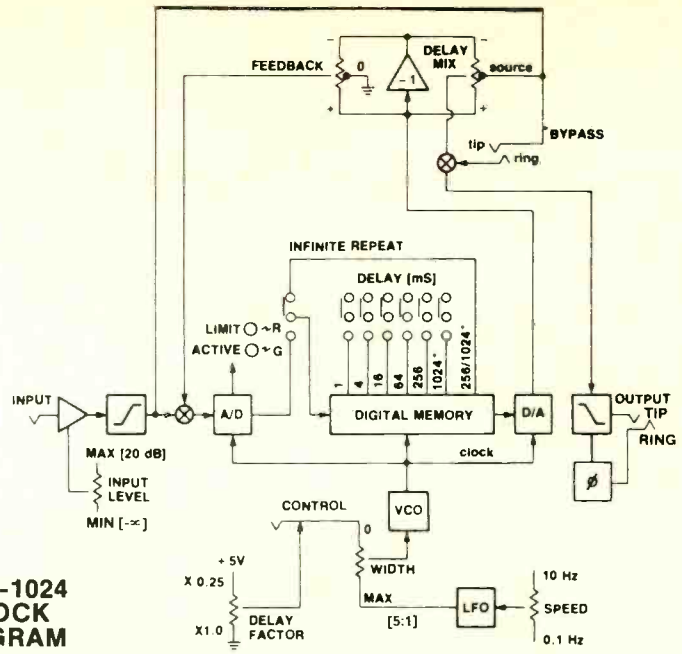
ing a sustained electronic screaming effect for a video graphics logo.

A borrowed "Lizard Man" synthesized voice was altered and lengthened using recirculating repeat echoes. The pitch was also changed via the modulation controls. When the desired sound had been recorded, it was transferred to an audio cart for routine insertion into daily network promos.

Infinite repeat echoes truly sounded perpetual. An increment of a swept tone along with a little out-of-phase feedback was captured and stored in full memory using the Infinite Repeat pushbutton. To create a composite sound effect for a precise video sequence, this repeating echo was faded in and out, and simultaneously layered onto the multitrack tape along with other special effects. About a half hour later, the now-forgotten repeating echo signal source was checked against the taped effect; it was virtually indistinguishable from the original recording.

Once a special effect, such as those described above, was achieved, the sound produced was stable and did not drift, even though at times careful adjustment were needed to attain the exact result. The fact that the Effectron II does not have a off/on power switch may be a large contributing factor towards maintaing this signal stability, as well as reliability.

Repeating a special effect after the controls have been reset proved difficult. Without numeric displays, delay times could not be accurately set back to their previous values. Although all the knobs have a line marker on each cap, there are no markings or pointers down the sides of the knobs to aid the user in recalling fine settings. The graduations on the panel around each knob are adequate, but labelled sparingly. For example, the Feedback control marking have only three labels, 0, (-), and (+), with four graduation marks between each label. At the end of a session, notation of precise settings onto track sheets for future reference became quite a tedious experience.



ADM-1024 BLOCK DIAGRAM

Summary

In spite of the device's few minor drawbacks, the Effectron II ADM-1024 offers excellent quality at the exceptional price of $499. DeltaLab also manufactures two other models in ths series: the ADM-256 and ADM-64. Both products provide all the same features mentioned above, except for smaller delay-time ranges of 256 and 64 milliseconds, respectively, and an envelope follower circuit on the ADM-64 that can be used to control flanging effects. Recommended retail prices are $399 and $299. ■■■

SUMMARY OF ADM-1024 SPECIFICATIONS

Delay Range:
Flange 0.25 to 16 ms.
Double 16 to 64 ms.
Echo 64 to 1024 ms.
Dynamic Range: 90 dB typical.
Frequency response: 20 Hz to 16 kHz, typical.
Distortion: 0.2% max.
Modulation:
Width 5:1
Speed 0.1 to 10 Hz.
Envelope Follower: No.
Feedback: Positive and Negative.
Bypass: Yes.
Infinite Repeat: Yes (Front & Rear).
Simulated Stereo Output: Yes.
External VCO Control: Yes.
Dimensions: 1¾ × 19 × 7 inches (HWD).
Weight: 10 pounds.
Price: $499.
Manufacturer: DeltaLab Research, Inc. 19 Alpha Road, Chelmsford, MA 10824. (617) 256-9034.

In the early evening of Sept. 17, 1973, Jay Barth was at the wheel of a 22 ft. utility truck that was loaded with sound equipment. Just south of Benton Harbor, MI an oncoming car crossed the center-line; fortunately Jay steered clear of the impending head-on collision. Unfortunately, a soft shoulder caused the truck to roll two and one half times. Exit several Crown DC-300A's through the metal roof of the truck's cargo area.

The airborne 300A's finally came to rest — scattered about in a muddy field, where they remained partially submerged for four and a half hours.

Jay miraculously escaped injury; the amplifiers apparently had not.

Unbelievably, after a short time under a blow-dryer all the amps worked perfectly and are still going strong.

The rest — and the truck, is history.

crown

1718 W. Mishawaka Road, Elkhart, IN 46517
(219) 294-5571

For additional information circle #91

New Products

ALPHA AUTOMATION ANNOUNCES BOSS 8400 AUTOMATION SYSTEM

The Boss™ is described as offering a new, comprehensive approach to the sound recording, editing, assembly, and audio-for-video post-production process. It combines the previously separate activities of sound synchronization, console automation, and other control room functions to provide intergral central control of, and communications between, virtually all studio equipment.

A commercial-grade 16-bit microcomputer and software to form the nucleus of a complete system, which will interface to mixing-console automation, programmable equalization, digital reverberation systems, MIDI, track select, the integration of production paperwork, and a Local Area Network (LAN) to tie together multiple control-room installations and the front office.

ALPHA AUTOMATION
2049 WEST BROAD STREET
RICHMOND, VI 23220
(804) 358-3852

For additional information circle #92

GOTHAM INTRODUCES SYSTEX COMPUTERIZED AUDIO RETRIEVAL SYSTEM

SYXTEX™ is described as the world's first fully computer-based audio system for film, broadcast, and recording studio use, and comprises a 330-megabyte hard-disk storage system that records, plays back and locates audio information with high accuracy and speed. The unit was created specifically for the needs of film/video editors and radio broadcasters who require

instant random access to a large library of high-quality audio material for dialog, music and sound effects, plus news broadcasts and commercial spots.

Recording engineers can utilize SYSTEX as a replacement for cumbersome sound effects libraries, and as a controllable command center to assemble and edit soundtracks. Access between selections on a disk is made without any audible interruption of the audio signal.

Heart of the system is a 16/32-bit Motorola 68000 microprocessor, with storage handled by Winchester-type SMD compatible 330-megabyte disk drive, or interchangeable disk-pack drives. For demonstration purposes at the recent NAB Convention, Gotham connected a Hewlett-Packard 9816 personal computer to operate the system and to show how easily it is programmed. (Any business computer can connect with the system via its RS-232C port.)

With the AES proposed standard sampling rate of 48 kHz, the unit can store and reproduce 60 minutes of mono recording time, or 30 minutes of stereo per disk. Its expandable architecture can accommodate as many as 60 separate disk drives, offering a storage capacity of up to 60 hours of mono program material.

Price for the basic SYSTEX dual-rack mounted system, including CPU, single hard disk and sequencer, is $125,000; additional sequencers are $35,000, and the 330 MByte Winchester disk drives are approximately $10,000 each.

GOTHAM AUDIO CORPORATION
741 WASHINGTON STREET
NEW YORK, NY 10014-2070
(212) 741-7411

For additional information circle #93

NEW MONITOR TECHNOLOGY FROM FOSTEX

The Pro Sound Division has introduced a new line of compact, point-source reference monitors, RM765/780/880, according to Ted Telesky, national sales manager. "These are the first monitor speakers to employ our RP Technology," Telesky said, "and we're very optimistic about this application."

The Fostex RP Technology was first expressed in the company's headphones, then microphones. The essential idea of the RP transducer is a flat, thin-film diaphragm onto which the voice coil is printed; this assembly is then suspended between magnetic circuits arranged on both sides of the diaphragm, with the same polarities facing each other and the opposite polarities adjacent to one another. Under this arrangement, the entire surface of the diaphragm is driven uniformly, Fostex claims, thereby avoiding phase anomalies and giving rise to the term RP, or Regulated Phase.

The Reference Monitor Series employs a coaxial design with a time-compensation adjustment such that the RP tweeter and double-spider woofer operate in a phase coherent, true concentric manner. The result is a compact, "near-field" monitor with point source depth and definition.

The RM Series consists of three models, all of which use the patented RP tweeter. Model 765 has a 6½-inch double-spider woofer; Model 780 an eight-inch double-spider woofer. Both of these models may be rack mounted with optional adaptors, and both models have a "field adjustment" switch for 2-pi or 4-pi environments. The third model in the series, RM 880, is currently under development, and will feature a 12-inch, three-way system utilizing the eight-inch coaxial assembly and a 12-inch edgeless woofer with a true linear motion of one inch, peak-to-peak.

Models RM 765 and 780 carry per-pair price tags of $438 and $598, respectively.

FOSTEX CORP. OF AMERICA
15431 BLACKBURN AVE.
NORWALK, CA 90650
(213) 921-1112

For additional information circle #94

NEW MATCHLESS CONSOLES FROM AMEK/TAC

The Matchless Series of consoles offer such standard features as full 24-bus routing, four-band EQ with swept mids and selectable bandwidth, eight sends, eight returns, eight mono sub-groups, two programmable mute groups, and line-return facility through the monitor section during the mix.

The new range is a scaled-down verison of the Angela Series console utilizing the same circuit design, and many of the same operational features, but in a smaller, more compact package.

The Matchless comes standard in a 26/24 mainframe configuration, with larger frames and retrofit extenders also available. An optional version containing a full patchbay and producer's desk is forthcoming.

Retail price for the standard Matchless 26/24 is $19,950; with patchbay, producer's desk and console stand will retail at $24,950.

AMEK CONSOLES, INC.
10815 BURBANK BLVD.
N. HOLLYWOOD, CA 91601
(818) 508-9788

For additional information circle #95

dynafex™...

HAVE YOU HEARD?

Available in rack-mount or modular formats.

The Dynafex noise reduction system has grabbed a lot of attention since its introduction, mainly because it provides up to 30 db of noise reduction without the encode/decode process. Impossible you say? Then you haven't HEARD the Dynafex.

We have been receiving rave remarks from engineers all over the world. Some of these comments have been, "It REALLY does work!", or "Best single-ended system I've ever heard!", or "Every audio facility should have one." Engineers have discovered that the Dynafex can be an invaluable problem-solver and can sometimes even be a client-saver.

Noise problems are encountered every day that conventional encode/decode systems or noise gates cannot deal with. Having the capability of being used on virtually ANY audio signal, along with providing a dramatic amount of noise reduction is what makes the Dynafex a vital tool in any audio facility.

Contact your nearest professional audio dealer for more information on this totally unique noise reduction system. You owe it to yourself...and especially to your clients.

Here's what the magazine reviewers have said:

"...Performs what might be called 'sonic miracles'..." Len Feldman, Modern Recording & Music

"I was most impressed with the Dynafex, and everyone I have shown it to has felt the same." Richard Elen, Studio Sound

"Its ease of usage, versatility, and effectiveness make the Dynafex most desirable." Robert Musso Music & Sound Output

Current Applications:

- During recording
- On-line in radio
- Concert sound reinforcement
- Mixdown recording
- Disc Mastering
- On-line in Television
- Tape duplication
- Video/Film production
- Cable television
- Telephone lines
- Satellite communications
- Motion picture sound
- Audio sweetening
- Microwave transmission

International Distributor:

J.C. Audio Marketing
(617) 368-0508
Telex: 951546 JCAUDIO

micmix

MICMIX Audio Products, Inc.
2995 Ladybird Lane
Dallas, TX 75220
(214) 352-3811

For additional information circle #96

New Products

STUDER TLS4000 AUDIO/VIDEO SYNCHRONIZER SYSTEM

For simple synchronization requirements, a single TLS4000 unit can control and slave one tape transport to SMPTE timecodes, reference frequencies, video composite signals, and biphase signals. The different operating modes can be accessed by the optional local control unit, which is equipped with timecode readout and control keys. Two different control panels, offering two levels of control possibilities, are available: one to satisfy the basic needs in synchronization, the other to provide the experienced operator with extended capabilities.

The system interfaces to all sorts of transports via an intelligent interface. For reference, it accepts timecode, frequency signals between 20 Hz and 20 kHz, video composite signals, as well as biphase signals. The synchronizer can operate in resolver or timecode interlock modes to satisfy a wide range of applications.

The basic TLS4000 can be operated as a stand-alone unit via the local control unit, or in large systems under SMPTE/EBU bus control via its serial port. By utilizing the serial control port (RS-232 or RS-422), a large number of synchronizers can be linked by a control computer, thus allowing synchronizer systems to be tailored to size according to specific facility requirements.

Features include the ability to fully synchronize timecodes with different standards (for example, 24 FPS code with 29.97 FPS codes); automatic switchover from master timecode to auxiliary reference after achieving synchronization, if so desired: resolver mode with selectable reference of standard timecode; slew mode for expanding or compressing time in interlock mode; and 10 cue and 10 offset registers.

The basic TLS4000 system (synchronizer module, interface, and local control unit) fits in a pair of 1-U rack spaces, and has a list price of $5,190.

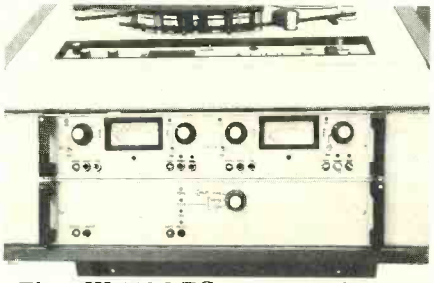
STUDER REVOX AMERICA, INC.
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

For additional information circle #97

MCI/SONY JH-110 WITH CENTER-TRACK TIMECODE

The new JH-110C-3-TC is a two-track capable of recording and reproducing timecode or pilot tone on a track in the center of ¼-inch tape. According to George Currie, division vice president and general manager, the new unit provides affordable audio synchronization by replacing the customary four-track machines used in post-production applications.

The machine's special heads make it possible to record as narrow strip of timecode or pilot signal in the guard band of the NAB standard two-track format. Precision engineering is said to eliminate crosstalk between audio and time code channels. Code can be recorded before, during or after audio, greatly simplifying production techniques involving synchronization. Although the code record and reproduce heads are physically separated from the standard audio heads, delay circuitry aligns all signals for razor-blade editing.



The JH-110-3-TC is available unmounted for standard 19-inch rack installation, or in variable profile (VP) or high profile (HP) cabinets.

MCI/SONY PROFESSIONAL AUDIO PRODUCTS
PARKRIDGE, NJ 07656
(201) 930-6137

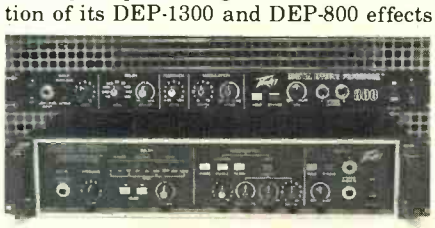
For additional information circle #98

PEAVEY LAUNCHES NEW DECA AMPLIFIER AND DEP SIGNAL PROCESSORS

Representing the second range of products developed at its recently completed AMR research facility, the new Digital Energy Conversion Amplification (DECA) power amps are said to exceed 90% in efficiency. Since the unit is highly efficient, large transformers and the associated circuitry is eliminated. The amplifier operates in a manner which minimizes the need for heat sinks, bulky power supplies, and other parts traditionally associated with high-power amplifiers.

After the initial introduction of DECA technology with the DECA-600 (300 watts per channel), the company plans to announce several higher powered versions. Peavey also plans to incorporate these new digital power amplifiers into its range of sound reinforcement products. The initial DECA-600 lists for $699, and should be available by mid-year.

Peavey also has entered the field of digital signal processing with the introduction of its DEP-1300 and DEP-800 effects



units. The DEP-1300 is a full 12-bit system, and enables full 20 Hz to 20 kHz performance with a maximum signal delay of 1.3 seconds. A company spokesman indicated that Peavey chose to go with a full 12-bit system to take advantage of the greater dynamic range capability over the majority of the competition's 8-bit systems.

The DEP-800 utilizes an 8-bit system, and is designed to compete in the lower price ranges.

Hartley Peavey, company president, indicated that these new digital signal processing units are the first of a wide range of products currently under development.

PEAVEY ELECTRONICS
711 A STREET
MERIDIAN, MS 39301
(601) 483-5365

For additional information circle #99

PANASONIC INTRODUCES RAMSA WR-8616 MIXING CONSOLE

The new WR-8616 is said to serve equally well for eight- and 16-track recording, as well as post-production sound applications. The unit is equipped with a variety of modules, including mono mike/line (WU-8101) and stereo line (WU-8106) versions. Depending upon which modules are used, the console can accommodate up to 16 mike-in, or 32 line-in signals, or a combination thereof. Two group output modules are offered: a basic unit (WU-8107) comprised of a fader, L-R send, and pan, solo, group on switches; and a tape monitor group module (WU-8102) which, in addition, features four channels of tape monitoring with both monitory (L-R busses), and send (L-R busses) level and pan controls.

Input channel signals can be routed through any one or combination of the group output channels to the L-R master output channels, or routed directly to the pan control. In broadcast applications, such a format allows up to three simultaneous stereo mixes. A mono ouput master provides a simultaneous mono output with the L-R master outputs.

Mike inputs of the monitor section may be employed without altering the patch connection, enabling recordings to be made while monitoring the actual input signals. All tracks going into the tape machines can be monitored while monitoring the actual input signals. Additional flexibility is featured through 16 insertion jacks on the input modules. Also, four return jacks are provided to connect a second mixing console, or to return a four-track recorder or effects to the group modules.

The unit's monitoring system permits direct monitoring of all 16 inputs. On a multitrack machine, monitoring of up to 16 channels can take place while a recording is in progress. The eight LED bargraph meters allow monitoring of channels 1 thru 8 or 9 thru 16, in addition to stereo monitoring of channels 13 to 16. The seven output signals — four group and left, right

and mono masters — are monitored with six VU meters.

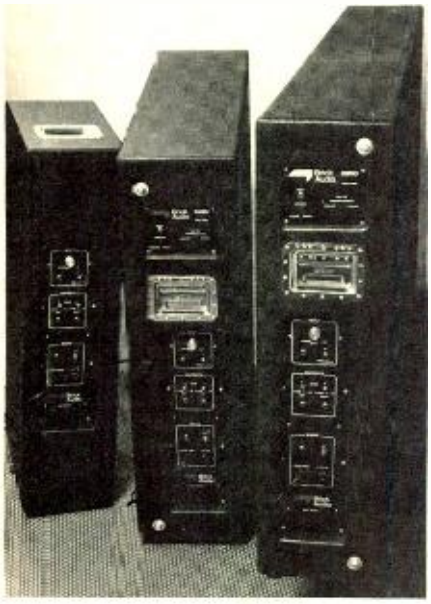
PANASONIC/RAMSA
ONE PANASONIC WAY
SECAUCUS, NJ 07094
(201) 348-7464

For additional information circle #100

BRICK AUDIO LAUNCHES NEW SERIES OF PLATE REVERB UNITS

The new plate reverbs are available in four sizes ranging from 1 by 3 feet, to 3 by 7 feet. Retail pricing begins at $699.

The units' drive section does not require a traditional voice-coil/magnet structure. Rather, it uses Brick's patent-pending solid-state transducer system, which eliminates the need for time-consuming alignment procedures, and is said to increase the signal-to-noise ratio to 80 dB or better.



The drive section of the unit features a "brilliance control" that enables the user to change the characteristics of the room being simulated from "warm dark" to "open and bright." The return sections feature adjustable controls (±10 dB) and are configured in a stereo return.

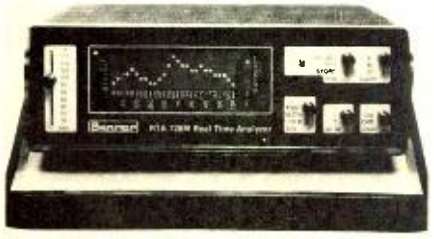
Motor drive damping is standard on three of the four units. A rack-mount remote control unit is an available option.

BRICK AUDIO
102 SOUTH PORTER
ELGIN, IL 60120
(312)742-7425

For additional information circle #101

BANNER REAL-TIME SPECTRUM ANALYZERS

Two new products, the RTA-728 and RTA-728M, feature battery or AC operation, and can be handheld or placed on a benchtop with the handle serving as an adjustable tilt stand.



Both models are 27-band third-octave analyzers with frequencies set to standard ISO centers. A broad-band sound pressure level display is also provided, and consists of a matrix of 196 LEDs with selection of 1 dB or 3 dB per display step.

Display decay rate can be set to Fast, Slow, or Hold. Balanced microphone inputs are on XLR connectors, and line inputs are on standard ¼-inch inputs. Double-tuned filters are standard.

The RTA-728M has all the features of the RTA-728 but with the addition of three non-volatile memories. At the push of a button, displays are stored in memory and can easily be recalled for use later.

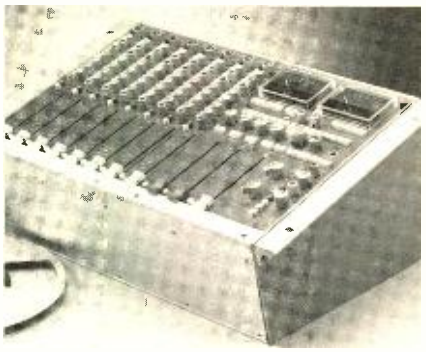
Suggested list price of the RTA-728 is $395; the RTA-728M (with memory) is $595.

BANNER
P.O. DRAWER 1803
SHELBY, NC 28150
(704) 487-7012

For additional information circle #102

CALREC INTRODUCES M SERIES PORTABLE MINIMIXERS

Two frame sizes are available to house between eight and 16 input channels, plus optional twin-channel compressor-limiters (with stereo-link switch). Features include 30mm modules for improved space saving and operator comfort; mike/line input selection with 10 dB per step switching; phase reversal plus high- and low-pass filters; three-band equalization per channel; four auxiliary outputs, each switchable pre/post fader; pan to stereo output; channel PFL and AFL; group or stereo A and B faders with PFL; mono master fader taking a mix of the A and B faders plus oscillator, talkback, 48-volt phantom power and headphone monitoring.



Options include: external circuit controls (for example, tape-machine remotes); PPM or VU meters; and an insert system from either pre- or post-channel fader and master fader.

Five different Minimixer styles are available, including a tabletop design with control surfaces at a shallow angle, mounted in a flight case, or in a 19-inch rack mounting version. The M Series powers from either a 240/120 AC source or from an externally supplied 16V DC.

AUDIO+DESIGN/CALREC, INC.
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #103

New Products

NEW NAGRA IV-STC PORTABLE WITH CENTER-TRACK TIMECODE

The new version of the IV-S enables the recording of a separate SMPTE/EBU timecode track for film and video synchronization. Size and presentation of the machine has not changed, and no external accessory is necessary for normal operation.

Mechanically the new machine is identical to the IV-S, the only difference being a thin sliding panel mounted underneath. The panel contains an LCD display and a mini keyboard used to read or enter data, and to check the operation of the system.

NAGRA MAGNETIC RECORDERS, INC.
19 WEST 44th STREET
NEW YORK, NY 10036
(212) 840-0999

For additional information circle #105

MODEL 5500 PROGRAMMABLE MEASUREMENT SYSTEM FROM AMBER

The new unit can be used to measure gain, frequency response, output power, weighted and unweighted noise, crosstalk, total harmonic distortion, intermodulation distortion, phase response, and wow and flutter. The benchtop instrument can

be controlled using a conventional front panel, an external controller for ATE applications, or the in-built non-volatile user programmability.

The instrument offers high performance with distortion to as low as 0.003%, and high speed with readings yielded in typically two seconds or less. It features fully balanced and high-level input and outputs.

The system contains a high-performance generator with output capability to over +30 dBm balanced. The analyzer can measure level in volts, dBV, dBm, or watts. Frequency is measured to five-digit accuracy, noise with four or eight selectable weightings, total harmonic and intermodulation distortion to SMPTE, DIN or CCIF requirements. Measurements can be true RMS, average, or quasi-peak. A built-in spectrum analyzer with four selectable bandwidths allows noise floor characterization and crosstalk measurements.

The user can store 10 complete instrument setups in non-volatile memory for rapid recall of repetitive tests. And the 5500 can be interfaced to virtually any computer with built-in serial RS-232 and parallel IEEE-488 interfaces.

Basic system price of the Model 5500 is $6,500.

AMBER ELECTRO DESIGN INC.
14125 CAPRI DRIVE
LOS GATOS, CA 95030
(408) 378-3501

For additional information circle #107

LYNMAR INDUSTRIES UNVEILS HALON FIRE EXTINGUISHERS FOR STUDIOS

The 1211/1301 Halon fire extinguishers create instantaneous action on fires, and require absolutely no clean-up. Unlike carbon-dioxide or dry chemical extinguishers that may permanently damage recording consoles, rack-mount equipment, tape machines, 1211/1301 will not damage equipment. The units are designed specifically for B- and C-type fires.

The four principles by which Halon works are envelopes the burning substance with ice; replaces the oxygen; removes the heat; and chemically breaks the fire chain in a matter of milliseconds.

Lynmar's 1211/1301 extinguishers come in a 2.7-pound unit; since it is charged with Halon (not nitrogen) it never needs re-charging. Fire Fighter is ARL approved. List price of the new 1211/1301 extinguishers is $79.95.

LYNMAR INDUSTRIES, INC.
962 SUNRISE HWY.
BABYLON, NY 11702
(516) 422-5111

For additional information circle #108

OTARI MODEL EC-101 SYNCHRONIZER MODULE FOR MTR-90 SERIES

When fitted with the EC-101 in-machine synchronizer module, the MTR-90 transport is capable of maintaining frame lock

in both forward and reverse directions from 0.2 to 2.5 times play speed, and will park within two frames of the master machine.

The EC-101 is the first in a series of syn-

chronizers specifically designed to optimize transport performance characteristics of all MTR Series machines.

The compact module plus into an open space in the MTR-90's electronics bay, and requires only timecode interface from the master — no tallies being necessary. Digital Widelok™ maintains phase lock of ±50 microseconds over a ±50% play speed range. Proprietary, gate-array-based LTC readers read from 1/25th to 60-times play speed (using timecode recorded at 15 IPS). Offset storage is in 1/80th frame increments.

Price of the EC-101 module is expected to be less than $3,500.

OTARI CORPORATION
2 DAVIS DRIVE
BELMONT, CA 94002
(415) 592-8311

For additional information circle #109

AUDIO+DESIGN/CALREC OFFERS "PRO" VERSION OF SONY PCM-701ES DIGITAL PROCESSOR

The company has developed three stages of "professionalization" for the Sony PCM-701ES. Section One is designed for use in the studio, and offers the convenience of a professional, electronically balanced audio interface, with a ganged 12-position switched input/output attenuator. Also included is the patented CTC circuit, and a 19-inch rack mount cradle.

Section Two adds a digital input/output interface for connection to future Audio+ Design developments. Also provided is switching for NTSC/PAL standards, and copy prohibit and record pre-emphasis on-off.

Section Three will include the PCM-1610 digital input and output interface, and remote switching of rear-panel function supplied in Sections 1 and 2.

A professional update also is available as retrofit for systems already purchased.

AUDIO+DESIGN/CALREC, INC.
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #110

TANNOY UNVEILS WILDCATS RANGE OF PA SYSTEMS

Many years of research are said to have gone into the development of the systems, and for the last two years certain drive units have undergone rigorous road use with bands such as Saxon and AC-DC, where they comprised the entire mid and ultra high-frequency sections of the main PA.

According to Tannoy, its research showed that many customers wanted systems which could grow with them and not become obsolete. The Wildcat's modular construction ensures that whether you use two or 22 enclosures the system adapts itself to the artists' needs. Where the venue or programme material calls for low bass response, the Leopard 15-inch bass system can be used. Unlike most so-called "modular systems", the unit does not necessarily need bi-amplification since it incorporates its own passive crossover enabling one amplifier to drive, for example, two Leopards and a Lynx dual 12-inch cabinet.

The new cabinets are aimed at three market segments: firstly, the group PA area where they should be the natural choice for cabaret artists who require clarity and flexibility from their system, as well as the ease of installation; secondly, the discotheque and club industry, to provide high levels of sound with characteristics normally associated with studio monitoring; and finally, audio/visual systems that require simple installation and guaranteed sound quality.

Recommended retail prices range from $898 for the Cougar stage wedge monitor, to $1,398 for the dual-12 Lynx cabinets.

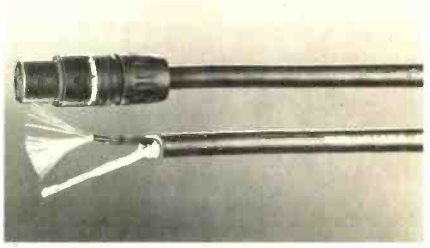
TANNOY-CROWN
97 VICTORIA STREET
KITCHENER, ONTARIO
CANADA N2H 5C1
(519) 745-1158

For additional information circle #111

MONSTER CABLE ANNOUNCES PROLINK MI AND MIKE CABLES

The three new Prolink™ Series MI models and two mike models are designed to ensure durability and withstand the rigors of heavy professional use. Fully shielded for ultra-low noise and hum, each cable is fully terminated with precision connected covered by scratch-proof black chrome plating.

Prolink Series 1 High-Performance microphone cables comprise large, multi-gauge conductors that are said to "align frequencies in both amplitude and phase as they travel through the cable, restoring lost clarity and warmth." Series 2 small-sized mike cable utilizes the same winding configuration as the Series 1, but in a more compact package.



Series 1 and 2 cable is available in 10-, 15-, and 20-foot lengths terminated in XLR connectors, or in bulk quantities.

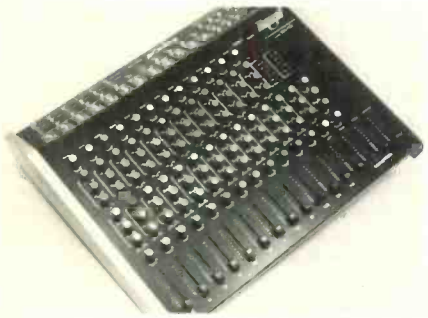
MONSTER CABLE
101 TOWNSEND STREET
SAN FRANCISCO, CA 94107
(415) 777-1355

For additional information circle #112

New Products

TANGENT SX SERIES STEREO CONSOLES

The SX Series of 8-, 12- and 16-input mixers feature three independent sends in addition to the stereo mix. Adding a separate effects bus to the usual reverb and monitor busses is said to give a sound engineer the extra flexibility of sending an *indpendent* mix to a flanger or digital delay line. Or, with a simple modification, the extra effects bus can be converted into an extra monitor output.



Professional-standard 48-volt phantom power also is included, plus three-band EQ on each input channel. Separate insertion jacks are provided for patching external effects into individual channels.

The SX-8 is a rack-mounting unity with the jacks located on the back of the unit for easy access; the SX-12 and SX-16 have a sloping rear panel for easy operation while in a road case.

Suggested list prices are $950 for the SX-8 rack, $1,200 for the SX-12, and $1,500 for the SX-16.

TANGENT
8206 N. DREAMY DRAW
PHOENIX, AZ 85020
(602) 997-4308

For additional information circle #114

NEW POWER AMP SERIES FROM TOA

Fully short-circuit-protected and current-limited, the five new amplifiers, Models P-75D, P-150D, P-300D, P-150M, and P-300M, are designed for stereo or mono operation, and contain speaker-protection relays plus front-panel circuit breakers. Each amplifier features cooling and heat sinks to ensure safe, stable operation. Signal ground and electrical ground may be separated.

In addition, green and red LEDS are provided on each amplifier's front panel for quick visual confirmation of signal presence, circuit protect/thermal warning, and output clipping.

Rear panels provide XLR and phone jack input wired in parallel and stereo models ("D") contain a rear-panel switch for the mono bridging mode. Each amp provides an octosocket for an input transformer if balanced input is required. Stereo models include high-pass filter on/off switches to prevent speaker damage from subsonic frequencies.

TOA ELECTRONICS, INC.
480 CARLTON COURT
SAN FRANCISCO, CA 94080
(415) 588-2538

For additional information circle #115

BEYER DT 108/109 HEADSETS

Because of numerous requests by intercom manufacturers and broadcast companies, the company has introduced a variety of new models of its DT 108 and DT 109 communications headsets.

The new models feature options like a balanced microphone line equipped with built-in amplifier to interface with carbon-mike inputs. Models are also available with a balanced mike line without amplifier.

Other options available include independent volume controls for the headphones and a "push-to-cough" switch.

In addition, Beyer Dynamic is offering more variety in terms of microphone and

headphone impedances so that the DT 108 and DT 109 can be correctly matched to any broadcast application.

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

For additional information circle #116

NEW RAMSA WX-8050 WIRELESS MICROPHONE FROM PANASONIC

The WX-8050 receiver is said to utilize a unique space-diversity system that ensures a reception stability at least two

times greater than comparable single-station reception systems. Two reception antennaes on stage receive the transmitted sound from the microphone, and the control section of the WX-8350 receive unit automatically selects the stronger transmission.

The receive circuit uses a crystal controller oscillator, fixed-frequency design which selects and receives up to three wavelengths designated in the UHF 400 MHz. The receive unit features a reception level meter with LED display for easier adjustments in dimly lit places, or from relatively distant points. Each of the twin receivers in the space-diversity system has green LED to show when it is active, and a red LED to show when it is inactive. A built-in squelch circuit suppresses EM interference such as car ignition, electrical appliance emissions, and fringe radio frequencies.

The WX-8050 wireless microphone uses a 24mm-diameter diaphragm, and derives continuous power for at least four hours from a single battery.

PANASONIC/RAMSA
ONE PANASONIC WAY
SECAUCUS, NJ 07094
(201) 348-7464

For additional information circle #117

BTX INTRODUCES "THE SYSTEM" SMPTE SYNCHRONIZER

The new multitransport interface/controller directly controls and synchronizes a master and slave in any combination of multitrack audio, video or film transport configurations. According to Michael L. Sipsey, VP marketing, The System™, priced at $7,995, represents a price/performance breakthrough by providing such advanced capabilities for transport autolocation, control and synchronization applications in a low-cost unit. The synchronizer will be available in July.

In addition to The System, the company has announced a Digital Control System (DCS)™ that is available in three distinct product models. The Model 7001 DCS is a Digital Multitrack System that includes two interfaces for Mitsubishi X-80 and X-800 transports, or the Sony PCM-3324, as well as the 3M M81, or any of the many U-Matic VTRs used for digital audio playback. Model 7002 DCS is a Digital Multitrack to Video Control System that includes the user's choice of digital multitrack and VTR interfaces. The third DCS model is a Digital Multitrack to Film control system (Model 7003 DCS) that includes a digital multitrack interface as well as a universal film dubber interface.

THE BTX CORPORATION
75 WIGGINS AVENUE
BEDFORD, MA 01730
(617) 275-1420

For additional information circle #118

ADA ANNOUNCES 2FX DIGITAL MULTI-EFFECTS PROCESSOR

The 2FX has a separate group of controls for each effect: flanger, chorus, and digital delay. The flanger or chorus mode can be used used simultaneously with the delay or repeat hold function. An exclusive patch switching system adds the flexibility of selecting the position of the effects in the signal processing chain.

The optional DM-2 footswitch controller provides remote access to bypass and effect selection. The DM-2's Memory Bypass™ function "remembers" any combination of pre-set effects. A single tap of the footswitch provides instant access to

the last combination of effects used or effect pre-set while in the bypass mode.

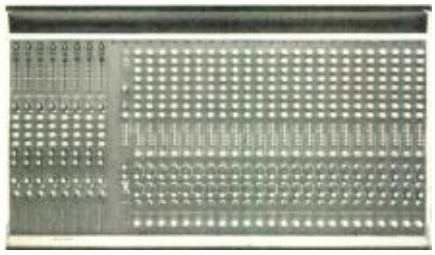
Other features include over one second of delay at 17 kHz bandwidth, and 10-to-1 sweep range (flanger). Suggested list price of the 2FX is $599.95; DM-2 footswitch costs $99.

ANALOG DIGITAL ASSOCIATES
2316 FOURTH STREET
BERKELEY, CA 94710
(415) 548-1311

For additional information circle #119

PULSAR ANNOUNCES M8 MONITOR CONSOLE

Main frame sizes range from a 16-input by 8-output, to a 48-input by 16-output. The M8 Series offers total modularity, including back panel, three-band sweep EQ, phase-reverse switch, mute switch, priority cue system, 48-volt phantom power, LED metering on all gain stages, two talk-



backs (house and stage), extensive headphone monitoring, and high- and low-pass filters, mute and cue. Also included is an additional four-send matrix on the input; these four sends per channel are sent to the four aux receives on the outputs and allow subgrouping of several channels together.

PULSAR LABS, INC.
3200 GILCHRIST ROAD
MOGADOR, OH 44260
(216) 784-8022

For additional information circle #120

NEW E-V SENTRY 100EL WITH INTEGRAL POWER AMP

The new Model 100EL combines the features of the Sentry 100A with a self-contained power amplifier. The monitor/amp combination weighs only 33 pounds, and measures just 17¼ × 12 × 11½ inches.

The 100EL features a 45 Hz to 18 kHz frequency response, with only a 3 dB variation. An eight-inch woofer is installed in

an optimally vented enclosure to provide the unusual combination of small size, extended bass response, and high efficiency. A Super-Dome™ tweeter is said to offer high frequency reproduction and uniform dispersion (120° at 5 kHz).

The Sentry 100EL can be driven by any line-level signal source — both XLR and ¼-inch phone jack connectors allow input from balanced and unbalanced lines. Further, the optional SRB-7 rack mount/-wall mount kit allows the unit to be used in virtually any monitoring environment.

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

For additional information circle #121

NEW STEREO AUDIO/VIDEO DELAY COMPENSATOR FROM LEXICON

The Model 1300 precisely compensates for video delay, holding lip sync regardless of what digital video processors are in use,

and compensates for satellite transmission delays. It decodes the hysteresis frame offset information from any video synchronizer for frame-accurate synchronization.

The stereo unit features full 16-bit digital audio processing. A microprocessor-controlled, removable delay configu[ration] control module can be software or h[ard]ware configured to conform to any delay sync decoding scheme. Three initial standard decoding options include a pulse-width, wild feed genlock, and a serial data. Front-panel switches can be used to set a delay offset value displayed in either milliseconds, or frame units. Stereo delay is from 0 to 340, 680, 1365 or 2048 milliseconds.

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

For additional information circle #122

OTARI PREVIEWS THREE NEW TAPE MACHINES

Unveiled at the recent NAB Convention in Las Vegas, product availability of the three transports is projected for the fourth quarter of 1984.

A new one-inch multitrack series, designated **MX-70**, was previewed in 8- and 16-track configurations. The machines feature switchable 15/30 IPS speeds, are active-balanced with +4 or -10 dB operating level, and have a wide range external VSO capability. They will come with a standard connector for interface to any SMPTE-based video editing system, machine controller, or synchronizer. RS-232C or RS-422 interfaces and half-inch/eight-track will be optional. Standard features also include spot erase, low-frequency equalization, record phase compensation, return-to-cue, and dump edit.

Designed for the broadcast market, a

Products

achine, designated **BTR-5**, ole in mono/full-track and o-track versions, with or channel timecode. The com- is fully microprocessor-controlled, and will accept up to 12-inch reels with constant tension reel motor servos. DC servo control of the capstan motor also is featured. Speeds are 15, 7.5, 3.75 IPS, and a three-location memory will be included. Optional equipment will include: RS-232C or RS-422 serial I/O port, autolocator, remote session controller, time-code head assembly, and pitch controller.

The new **MTR-20** series features computerized self-alignment, and will accept up to 14-inch reels. It will be available in both ¼- and half-inch formats, including half-inch/two-track and ¼-inch with center-track timecode.

OTARI CORPORATION
2 DAVIS DRIVE
BELMONT, CA 94002
(415) 592-8311

For additional information circle #124

SONY ANNOUNCES EIGHT-CHANNEL DIGITAL MIXER

The new eight-input mixing system is modular in design, consisting of a rack-mountable signal processor unit, desk-top control unit, and A-to-D and D-to-A conversion units, both rack-mountable. Typical applications would include CD mastering, digital mixing, and classical music location recording.

The system's processor unit connects directly to Sony digital equipment, and is designed to mix 16-bit linear quantized digital signals. The unit accepts eight digital input channels and provides three separate digital outputs: a two-channel main output, two-channel sub-output, and eight-channel direct output. The utilization of the control unit with the processor permits up to eight digital signals to be mixed.

The A/D and D/A conversion units provide variable headroom control and selectable input and output sensitivities.

Normal mixing console functions are provided by the control unit, including four-band EQ, low- and high-pass filters, PFL, pan, and LED peak program level displays.

MCI/SONY PROFESSIONAL AUDIO PRODUCTS
PARKRIDGE, NJ 07656
(201) 930-6137

For additional information circle #125

TWO NEW COMPACT PA MONITORS FROM JOE'S SOUND

The new mini monitors are said to combine high output and precise acoustical alignment with remarkably small dimensions. The 12mm and 15mm designs are available loaded with the customer's choice of JBL or PAS drivers and tops.

Cabinets are fabricated from ¾-inch, void-free birch plywood, and finished in satin black or oil and walnut stain. Mini monitors are also available in the company's specially reinforced resin/wood alloy compound for users who require maximum efficiency and least coloration of sound.

Dimensions of both units are: 12mm, fully loaded, 26 by 14 by 13, 53 pounds; and 15mm, fully loaded, 32 by 18 by 17, 65 pounds.

JOE'S SOUND & SALAMI CO.
981 SOUTH BROAD STREET
TRENTON, NJ 08611
(609) 394-5637

For additional information circle #126

BIAMP LAUNCHES 24 AND 28 SERIES MIXING CONSOLES

Featuring state-of-the-art design and construction to assure low noise, high slew rate and ultra-low distortion, the two new mixers are said to achieve performance levels comparable to the finest studio consoles in a compact, affordable package.

The 24 Series, with four-submaster mixing in stereo and mono, is available in 8-, 12-, 16- and 24-channel versions; and the 28 Series, with eight-submaster mixing in stereo and mono, is available with 12-, 16-, 24- and 32-channels.

Features include low-noise effects buses switchable pre or post; three-band EQ with sweepable mid-range; stereo monitors selectable for live or recording; channel solo switch; green/red LED input indicators; mike-in/line-in/tape-in/direct-out switching; flourescent metering; internal or optional external power supply; and 48V phantom power available on all inputs.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

For additional information circle #127

SCREEN SOUND PREMIERES VIDEO STREAMER FOR ADR/LOOPING

Like an "electronic grease pencil", the new unit superimposes a vertical bar at either side of the screen, and wipes one of the bars across the screen until they meet and disappear at the cue point. Front-panel controls allow fine adjustments of cue timing and manual triggering for set-up.

The Video Streamer takes its trigger-pulse from the Audio Kinetics Q.Lock or BTX Softouch beep-tone relay contacts, or any unused relay contacts on the Adams-Smith, MQS, Sony or other device. The Streamer will trigger from the first pulse only, and will ignore subsequent pulses for three seconds.

The front panel has two thumbwheel switches that allow precise and independent control of delay before the start of a wipe, and the time to wipe across the screen; each is adjustable from 0 to 10 seconds. The Streamer bars are *added* to the video signal and there is said to be no degradation of picture quality; full bypass is provided when the unit is not in use or powered down. The device is PAL- and NTSC-compatible, and does not interfere with vertical interval information.

The Video Streamer is priced at $1,350.

SCREEN SOUND, INC.
4721 LAUREL CANYON BLVD. #103
NORTH HOLLYWOOD, CA 91607
(818) 761-0323

For additional information circle #128

LOFT SERIES 600 FREQUENCY DIVIDING SYSTEMS

The new units provide, for the first time, the company says, *switchable* 18 and 24 dB slopes at all crossover points. Design engineers concluded that the ideal dividing network could not be 18 *or* 24 dB exclusively. Subtle differences in speakers, room conditions and speaker placement, it is claimed, make it necessary for both slopes to be available.

Features include electronically balanced inputs and outputs via locking phone connectors as standard. (XLR connectors are available at no additional charge.) Mounting bosses have been installed to accommodate transformers such as Jensen JE-123-SL.

Plexiglas security panels with special tamper-proof hardware and drivers are available as optional equipment.

PHOENIX AUDIO LABORATORY, INC.
91 ELM STREET
MANCHESTER, CT 06040
(203) 649-1199

For additional information circle #129

news

— *continued from page 36 . . .*

PANASONIC/RAMSA SOUND SYSTEMS FOR OLYMPICS

In making the announcement, Harry L. Usher, Los Angeles Olympic Organizing Committee executive vice president and general manager, said that the company had been chosen after a project team visited each Olympic venue to assess the acoustic needs, and to design appropriate sound systems.

"A complete custom package from RAMSA, including audio mixers, speakers, microphones, amplifiers and accessories will be provided," said Nick Hudak, RAMSA project manager. He added that all installation, maintenance and operation of the equipment will also be provided by Panasonic/RAMSA.

The 1984 Olympic Games will be held at more than 30 locations throughout Southern California. One of the more elaborate sound systems will be in the Los Angeles Memorial Coliseum, where a RAMSA WR-8616 house-mix console will provide outputs to 33 WP-9210 power amplifiers feeding a large array of RAMSA speakers, horns and drivers.

Among the audio equipment being supplied by Panasonic for the Games are 57 audio mixing consoles, 215 power amplifiers, an assortment of 900 speakers, 250 microphones and accessories such as equalizers, turntables, tape machines, cassette tape recorders, and special-effects equipment.

RAMSA sound systems installed at the Coliseum and at East Los Angeles College will be permanent systems that will remain in place after the Games. Other venues will have temporary systems for use only during the period of Olympic activities.

NEW DISTRIBUTION COMPANY SET UP TO HANDLE EUROPEAN PRODUCTS

Audiotec, Inc. is the name of a recently formed US distribution company for pro audio equipment, located in Hawthorne, New Jersey. According to Wayne A. Morris, sales and marketing manager, Audiotec is the parent company for a wide range of products, including Dynamix D3000 Series consoles equipped with 16 inputs and eight subgroups, and 32/16 formats for 16-track recording; Mitec monitor consoles with 16 inputs and eight subgroups; ACE 16/8 and 24/8 consoles and MFB 2005 Electric Drums and Digital Drum Computer with MIDI interface.

A second company, StarSound Dynamix US Distribution, which will function as distributor for Dynamix products in the US, is a joint venture between StarSound UK, the London-based manufacturer of Dynamix consoles, and Musik Productiv Germany. General manager Manfred Kayser states that StarSound Dynamix will be run as a separate division of Audiotec.

Audiotec, Inc. is located at: 96 Lafayette Avenue, Hawthorne, NJ 07506. (201) 423-4404.

AUDIOTECHNIQUES OPENS PARTS DEPARTMENT; FORMS A/T SCHARFF RENTAL FACILITY

James Gillespie is the new manager of the Audiotechniques parts and accessory department at the company's Manhattan facility. According to Gillespie, a feature of the new operation will be over-the-counter availability of parts for MCI recorders and consoles. Additionally, more than $100,000 has been spent for initial inventory of standard parts such as connectors, cable, tools, patch pays, cords, etc. Gillespie also said that his tape department has the largest inventory of Sony digital tape in the US.

Also, Geoff Hillier has been appointed technical services manager at Audio Technologies. Hillier, formerly director of engineering for Trident, USA, will have management responsibility for installation, service and equipment maintenance and parts at the firm's newly enlarged and remodeled New York City facility in the Brill Building, 1619 Broadway.

• In mid-April, the rental division of Audiotechniques joined forces with Scharff Communications to become A/T Scharff. The joint venture is described as the largest audio rental facility in New York, and probably the country. The new company will be managed by Scharff's management team, with Josh Weinberg serving as VP and general manager.

According to company president Peter Scharff, "The combined inventories of Scharff and Audiotechniques is massive. But, having the largest inventory is only half of it; getting it there on time, up and working is the other half."

For digital recording, the company already has a Sony PCM-1610 processor complete with Sony BVU-800 VCRs and DAE-1100 editor, as well as PCM-F1 portable systems. It will also specialize in MIDI computer programming, and computer-based synthesizers.

A/T Scharff is located at: 1619 Broadway, New York, NY 10019. (212) 582-7360.

AUDIO INTERVISUAL DESIGN APPOINTED REP FOR SONY DIGITAL PRODUCTS

The new company, headed by LA Record Plant president Chris Stone, has non-exclusive jurisdiction of the 11 western states, and has already delivered two Sony PCM-3324 digital multitracks to Neil Young and two PCM-3324s to LA's Village Recorder (see "Studio Update" for further details).

"The scope of our business goes far beyond the sales of equipment," Stone

emphasizes. "We formed this company to act as systems facilitators. Because of my involvement in the record, video and film industries I saw a critical need to synthesize systems which integrate the very latest equipment."

The new company intends to put together packages for film scoring that utilize Sony video-based systems. "We are creating ways for studios to start making money again," says Jim Pace, vice president of AID, and formerly managing engineer with Digital Magnetics. "Digital is here to stay," adds Pace, "especially for the film and TV industries. The first people to be involved will be the first ones to profit."

As part of the agreement with Sony, AID also will operate as an OEM for various Sony products. A new video-based film scoring system has been developed in cooperation with Roy Prendergast's Music Design Group, a leading film/video music editorial facility. The VideoScore system will be marketed worldwide, and incorporates Sony VCRs, monitors, large-screen projectors and microcomputers. (See feature article detailing recent advances in film-sound techniques elsewhere in this issue.) AID has developed the proprietary interfaces linking SMPTE time-code with computer data. "Our system will allow the small studio to get into the scoring business for about $25,000, whereas a film chain runs about $150,000," explains Stone.

Full details from: AID, 8474 Third Street, Los Angeles, CA 90048. (213) 853-02400.

QUAD EIGHT/WESTREX CONSOLES FOR TBS, MGM, TRANS/AUDIO AND TODD-AO

The company has delivered the second of two custom film re-recording consoles to The Burbank Studios. Designed by QE/W engineering staff to TBS' specific requirements, the two film mixing boards feature 54 inputs, three groups of six-track mixing buses with re-assigns, and are designed for six-track stereo remixing.

Currently in production for delivery this summer are some of the largest console systems ever undertaken, including a 72-input, 24-track film recording system for Trans/Audio, New York. In addition, a 164-input console being built for MGM's Main Theater is probably one of the largest audio control system ever designed. The major problem, outside of the sheer scale of the system, QE/W says, was "folding" the system to fit into its 25-foot mainframe.

The Trans/Audio and MGM consoles are designed to facilitate the use of three independent, interlocked six-track master recorders — one each for dialog, music and effects — as well as a composite mono recorder.

All four of the above systems incorporate QE/W's Compumix IV automation, a minicomputer-based system that uses the company's digital fader, and features an 80-megabyte hard-disk storage system.

Also in production is a second console for Todd-AO: a 54-input, six-track film board, also designed for three-recorder operation.

SOUNDCRAFT OPENS NEW YORK OFFICE

Effective June 1, 1984, Betty Bennett, general manager of Soundcraft's US base in California, will be temporarily relocating to Manhattan to establish the new facility, and to serve local accounts. The move, according to Philip Dudderidge, Chairman of Soundcraft Electronics, Ltd., "is necessary because of [our] newest project, the TS-24 in-line console, and Soundcraft's emergence into the broadcast market with the SAC-2, previewed this spring at the NAB in Las Vegas. With the large number of Soundcraft consoles being sold into the New York market, a local sales/service facility is a necessity." Soundcraft is now interviewing for a local salesman to fill the field sales position in the new location. Contact Betty Bennett at (212) 315-0877.

The new facility will be located at: Soundcraft Electronics, New York, 44 West 62nd St., #20C, New York, NY 10023. (212) 315-0877/315-0878.

FIRST AMBISONIC SURROUND-SOUND BROADCAST AT KWMU-FM IN ST. LOUIS

On May 12, KWMU-FM, the non-commercial station of the University of Missouri/St. Louis, became the first station in the US to broadcast Ambisonic surround-sound material. Two half-hour programs encoded in UHJ two-channel format were aired, and featured a variety of Ambisonic musical and spoken-word examples. The musical selection consisted of live jazz and classical recording made by KWMU, in addition to a wide range of commercial UHJ-encoded vinyl pressings and Compact Discs imported from England.

Listeners were invited by KWMU to hear the decoded programs at several hi-fi shops in and around the St. Louis area. The station also arranged for a demonstration to be given following the airings to the audience gathered to listen on the University campus. The demonstration comprised a short history of Ambisonics technology, and explanation of the Calrec Soundfield microphone and its control unit, UHJ transcoder and several decoders. All Ambisonic equipment for the recording, production and decoding of the broadcasts was provided by Audio+Design/Calrec. [For a full rundown of Ambisonic recording and broadcast techniques, see the December 1983 issue of *R-e/p. — Editor.*]

When planning the broadcasts the assumption was made, KWMU staff explains, that there would be few, if any, Ambisonic decoders in the station's coverage area. To enable the local audience to hear the transmissions Ambisonically, decoders and replay equipment were set up at several locations throughout the area. Five Minim Electronics AD10 and AD7 decoders were brought in specifically for the project.

Not having access to a four-track of its own to record B-Format outputs from the Calrec Soundfield control unit, KWMU personnel were forced to restrict their efforts to stereo UHJ recordings. The output of the Soundfield control unit was fed to an Audio+Design Propak II, and then to either a Sony PCM-F1 digital processor or a Technics RS-1500 reel-to-reel with Dolby-A noise reduction. The stereo sound was, as it was later found to be in the Ambisonic format, clear, clean and very "natural", capable of capturing great detail.

Most of KWMU's Ambisonic recording were of live concerts, and included solo classical guitar, jazz duo, big band, chamber orchestra, and solo organ to full orchestra. The performances took place in many different halls, some with fine acoustics and some with less so. Again, all were stored on the F1 or RS-1500 with Dolby-A. The commerical UHJ-encoded releases were from a number of labels, and offered a slightly wider selection. There was music of Sir Edward Elgar, as well as pieces for

brass and early music ensembles that were augmented by disks containing bluegrass, dixieland, modern big band arrangements, and popular music. An Ambisonic "sampler" was included, as were two very notable disks: a live recording of the Royal Choral Society and Band of the Grenadier Guards, and a KPM library music album demonstrating the first use of the new Ambisonic Pan/Rotate processor.

Receiving the largest turnout of any of the listening sites, about 60 people travelled to the University of Missouri. During the broadcasts the audience was free to move around to judge imaging and the Ambisonic effect at different positions in the listening room. The most impressive examples, station staff tell us, seem to have been Fellgett and Barton's recording of Hubert Parry's *Jerusalem*, where the listener is surrounded by the audience as it first rehearses and then performs with the on-stage chorus and band; and program host Joseph O'Connor's stroll around the microphone, followed by his stirring performance of Leo Sowerby's *Fanfare*.

A large majority of listeners attending the different sites were said to be highly enthusiastic about Ambisonics. "The feeling of being in the audience was very good. I definitely prefer it to stereo," and "when I closed my eyes I could imagine being at a live performance" were typical listener comments. KWMU-FM will continue to offer listeners a chance to evaluate Ambisonic surround-sound technology by broadcasting in their entirety the half-dozen or so UHJ-encoded concerts used during the recent transmission.

COMPUSONICS INTRODUCES DIGITAL MULTITRACK RECORDER/MIXERS

Scheduled for delivery this summer, the new Studio DSP-2000 Series is a computer-based system configured as a single-user workstation and designed for digital recording and mixing of live music in a variety of digital formats. The system is believed to be the first microcomputer system for all-digital signal processing, from microphone inputs to final mixdown.

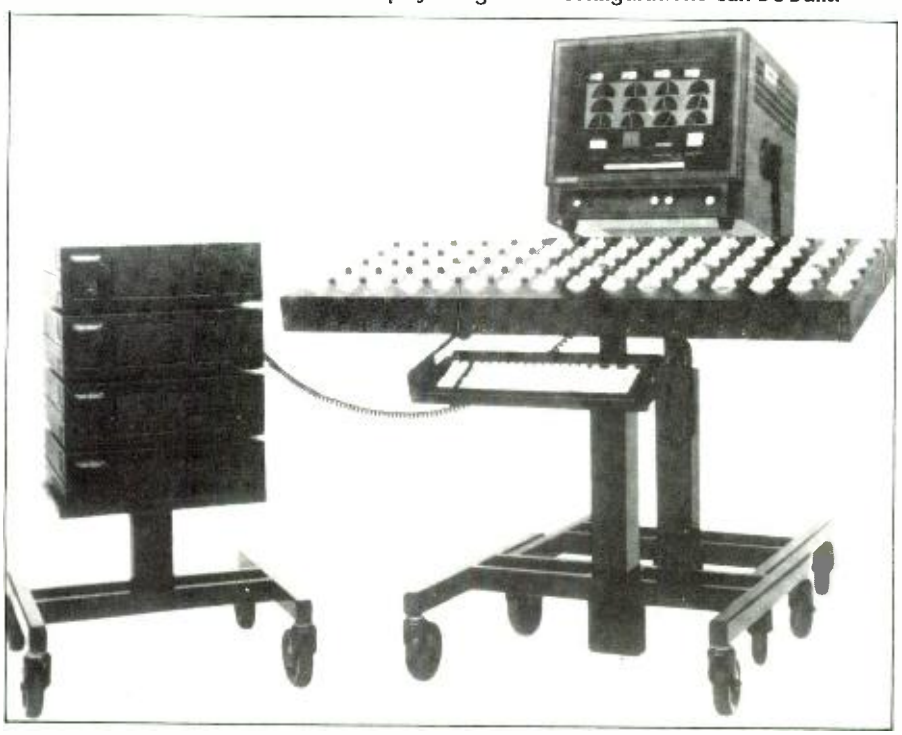
Sampling frequency for the A-to-D and D/A converters is 48 kHz, and all internal processing is 16-bit linear PCM. The system's mixer section features full EQ, cue sends, and monitoring outputs, while the recording section is based upon a newly developed, ultra high-capacity 5¼-inch floppy-disk drive said to be capable of storing 3.3 megabytes of digital data. (Currently under development are second-generation floppy drives that will be capable of holding 6.6 megabytes and, by late-1985, in excess of 20 megabytes.)

According to company president David Schwartz, the DSP-2000 Series will be made in four-track modules, and that the initial four-input/four-track systems will sell for about $30,000.

The system is comprised of a Motorola 68000 microprocessor-based CPU, color video monitor, full-stroke keyboard, trackball-arrayed control panel, printer, storage modules, signal processors, and input/output module — all of which is configured as a single workstation.

Digital processing is handled by a pair of TI TMS320 chips connected in series (two per channel) and running 32-bit arithmetic at a 20 MHz clock rate.

The new Compusonics Studio DSP-200 Series digital mixer and floppy-disk recorder comprises three units configured as a single user workstation: modular recorder, central "control surface," and color video display. Larger size configurations can be built.



news

The system is configured to run under the UNIX operating protocol, and utilizes C-based software.

"The Studio DSP-2000 Series will facilitate the production and supply of true digital master recordings for the first time," Schwartz asserts. "The system is capable of digital master recording in the standard Compact Disk format, as well as in the floppy-disk format for use in our consumer CompuSonics system."

Coincident with the introduction of the Studio DSP-2000 Series, the company also plans to deliver its first consumer system during early 1985. The DSP-1000 floppy-disk digital audio recording/playback system will allow consumers to make recordings from any digital or analog source. The consumer system allows for up to one hour of recording time at a 44.1 kHz, 16-bit sampling, and has the capability to receive digital transmission of music from remote data bases, and can interface with an IBM PC.

Further details can be obtained from Compusonic Corporation, 323 Acoma Street, Denver, CO 80223. (303) 698-0060.

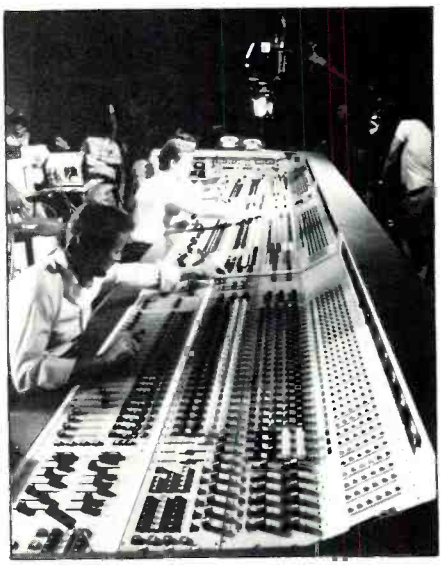
GLEN GLENN DEMONSTRATES CAPABILITIES OF ALL-DIGITAL FILM SOUNDTRACK

Digital Dream, a promotional film with an all-digitally recorded six-channel soundtrack, was recently screened in Hollywood before producers, directors, and entertainment industry production personnel. The project is said to represent a major commitment to digital audio by Glen Glenn Sound, one of Hollywood's largest film and video recording centers.

The debut screening of the 30-minute film, produced by The Pathelogical Corporation as a demonstration of Glen Glenn's digital capabilities, formed part of the ceremonies inaugurating a multi-million-dollar expansion of the company's facilities.

The plot of *Digital Dream* depicts the visions of three members of a re-recording team — a side of the film sound industry rarely seen by the public. To demonstrate the differences offered by digital audio, Glen Glenn replaced analog recording techniques in every phase of soundtrack production. "This is a landmark in the motion picture business," states Dana Wood, Glen Glenn's chief engineer. "Other productions have incorporated digital at various steps of the recording process, but we have used digital in *every* step".

"Digital audio is really the star of this film," Wood adds. "The experience of a digital soundtrack is like moving from the rear of the theater to the center of the action. Every sound effect has a crisp identity; every musical instrument stands by itself — and if silence is intended, what you get is *pure* silence."



Sound and camera crew prepare to shoot a scene at Glen Glenn Studio 1. Actors depicting the music, dialog and effects re-recording engineers are about to realize their "Digital Dream."

For the music elements, composer Ron Jones worked with former Bee Gees producer Albhy Galuten to create a blend of orchestral music with digital synthesizer effects. Twin Sony PCM-3324 digital multitracks were synchronized via a custom BTX Shadow system for 48-track sessions at the Glen Glenn/Record Plant scoring stage on the Paramount lot in Hollywood.

Back at Glen Glenn's new facility, dialog and sound effects also were recorded on digital multitracks. Sound effects not reproduceable in the studio — such as jet flyovers — were recorded on location using Sony PCM-F1 portable digital processors and U-Matic VCRs.

The second phase of the soundtrack production called for editorial decisions to be made for choice of tracks and exact synchronization with picture. Sony PCM-3324 multitracks were interlocked with Glen Glenn's computerized PAP (Post Audio Production) system as director and engineers viewed a SMPTE timecoded video workprint of the film. In the third stage of production, premixes were recorded to combine hundreds of tracks for the separate music, dialog and sound effects sub-masters.

"Digital audio has tremendous value in the pre-mixing stage," Wood explains. "It's not uncommon to go through three or four transfers, which [with analog] causes multiple noise generations. Noise reduction counteracts this effect in our usual operation, but with digital audio there is absolutely *no* degeneration of signal at all."

Following the pre-mix stage, the digital pre-dubs were taken into Glen Glenn's dubbing theater to produce the master six-track digital soundtrack mix.

For private screenings at Glen Glenn, a Sony multitrack is interlocked with a 70mm print from the 35mm anamorphic camera original to provide pure digital reproduction. Glen Glenn also has pre-

pared 35mm Dolby Stereo optical prints for screenings in conventional theaters.

"The soundtrack would have been absolute mush if we had recorded in analog," comments director/producer Eric Stahl. "For certain segments we combined as many as 200 tracks. Digital gives you a greatly increased dynamic range with incredible clarity and separation of the individual sounds.

"We are exploring the use of of digital technology in every phase of our operation," says Glen Glenn president Tom Kobayashi. "The major thrust of all our research and development is aimed in this direction. We have effectively demonstrated that digital recording can replace every analog soundtrack procedure. All editing and synchronization

. . . continued on page 170 —

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— *continued from page 167 . . .*

was done electronically. While we don't yet have digital playback in theaters, the immediate value is that a digitally-produced soundtrack can be transferred easily to any of the standard release formats, creating a vastly improved finished product."

"The Dolby print is outstanding, but the digital version is astonishing," concludes Koyayshi. "You can clearly distinguish the whoosh of a jet canopy being released while a full orchestra is reaching a dynamic crescendo. Digital audio provides the ultimate in clean, bright, and distortion-free sound."

OTARI AND DOLBY SIGN LICENSING AGREEMENT

Mike Pappas, national sales manager for the Industrial Products Division of Otari Corp., announces that an agreement has been signed with Dolby Laboratories Licensing Corporatin for Otari's use of the proprietary Dolby HX-PRO™ headroom extension process.

The HX-PRO circuitry enables high-level, high-frequency program material to be recorded more accurately, without sacrificing signal-to-noise ratio. Additionally, side effects of tape saturation, such as distortion, are said to be reduced.

The Dolby circuitry will be available for Otari's high-speed duplicating systems, Models DP7000 (64:1, 240 IPS) and the newly introduced DP80 System (64:1,480 IPS with 7.5 IPS masters).

Retrofits are anticipated to be available within 90-120 days for all applicable Otari machines.

STUDER BOOSTS SALES TO BROADCAST TV AND TELEPRODUCTION

Thomas E. Mintner, Director of Studer Products at Studer Revox America, noted recent sales of five 169/269 Series consoles to AME in Los Angeles, seven to NBC in New York, and WYES TV in New Orleans. All consoles will be used in video editing suites, or audio production.

In addition, NBC has purchased several Studer A810 audio recorders with center track SMPTE timecode for use in the network's New York video editing rooms.

Editel in Los Angeles has followed the lead of Editel New York with the recent purchase of a Studer A800 16/24-track and an A80 four-track, both of which will be used in the facility's video sweetening room. Editel New York recently added a Studer A810 timecode version to its new sweetening room.

MARTIN AUDIO VIDEO FORMS RENTAL DIVISION

According to Martin VP and general manager, Courtney Spencer, "Rentals are very important in the New York area, and we felt that the market could easily sustain another audio rental house." Spencer also announced the appointment of Jim Flynn to head the new Rental Division. Flynn is well known to studios and other equipment users in the Northeast from his two-year tenure as rentals manager for Audiotechniques.

The Martin Rental inventory will include the full range of recording, production, post-production, and broadcasting equipment, from microphones and mixers of all sizes to signal processors, monitor loudspeakers, and tape recorders of all sizes and formats from cassette to two-inch 24-track. Specially featured in the equipment listings are the full range of Otari tape machines; a wide variety of digital reverb units, including models from AMS, Eventide, Lexicon, Quantec, and Yamaha; two-channel digital recording processors from Nakamichi and dbx; and a choice selection of vintage, tube condenser microphones.

Contact Jim Flynn at (212) 541-5900, or write to Martin Audio Rental Division, 423 West 55th Street, New York, NY 10019.

BTX AND HARRISON SYSTEMS ANNOUNCE OEM AGREEMENT

The two companies recently signed an agreement covering the development of BTX automation products to be incorporated into the Harrison consoles. Under the agreement Harrison will offer an in-console version of the BTX Softouch™ audio editing system in its line of consoles. BTX will specifically tailor Softouch to Harrison's specifications, and the system will interface directly with console functions.

The Softouch system automates and controls many of the often disjointed post-production processes associated with sound-effects assembly, sweetening, dialog replacement, and Foley, and also provides timecode synchronization and control of multiple audio, video, and film transports.

Eric Johnson, Harrison's national sales manager, says that "With BTX capability, the Harrison Console System becomes virtually a cockpit command center for the user's mixing environment. The international reputation of BTX as the leading synchronization and timecode equipment supplier is what brought my attention to them for this project, initially. The Softouch System offers a great deal of flexibility and programmability unparalleled in any other product."

Michael L. Sipsey, BTX's vice president of marketing describes this venture as, "An industry first step to provide more cost-effective, user-friendly, system-oriented solutions to our customers."

The BTX Softouch System will continue to be sold as a stand-alone system directly through the company's dealer and distributor network. ■■■

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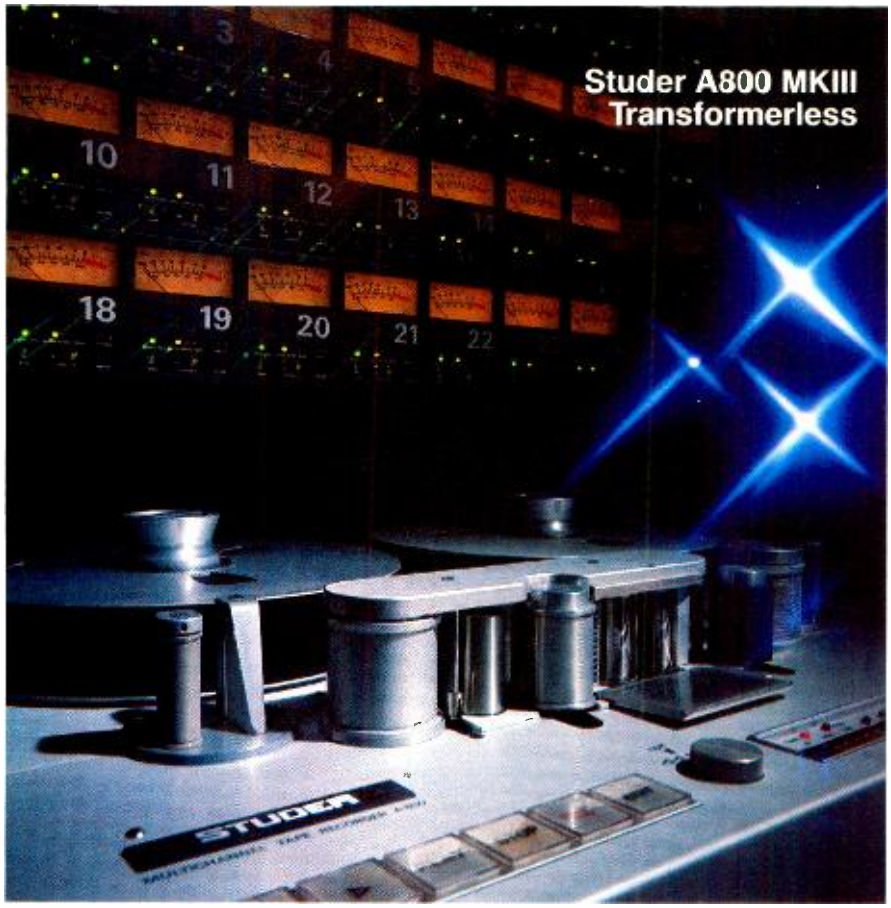

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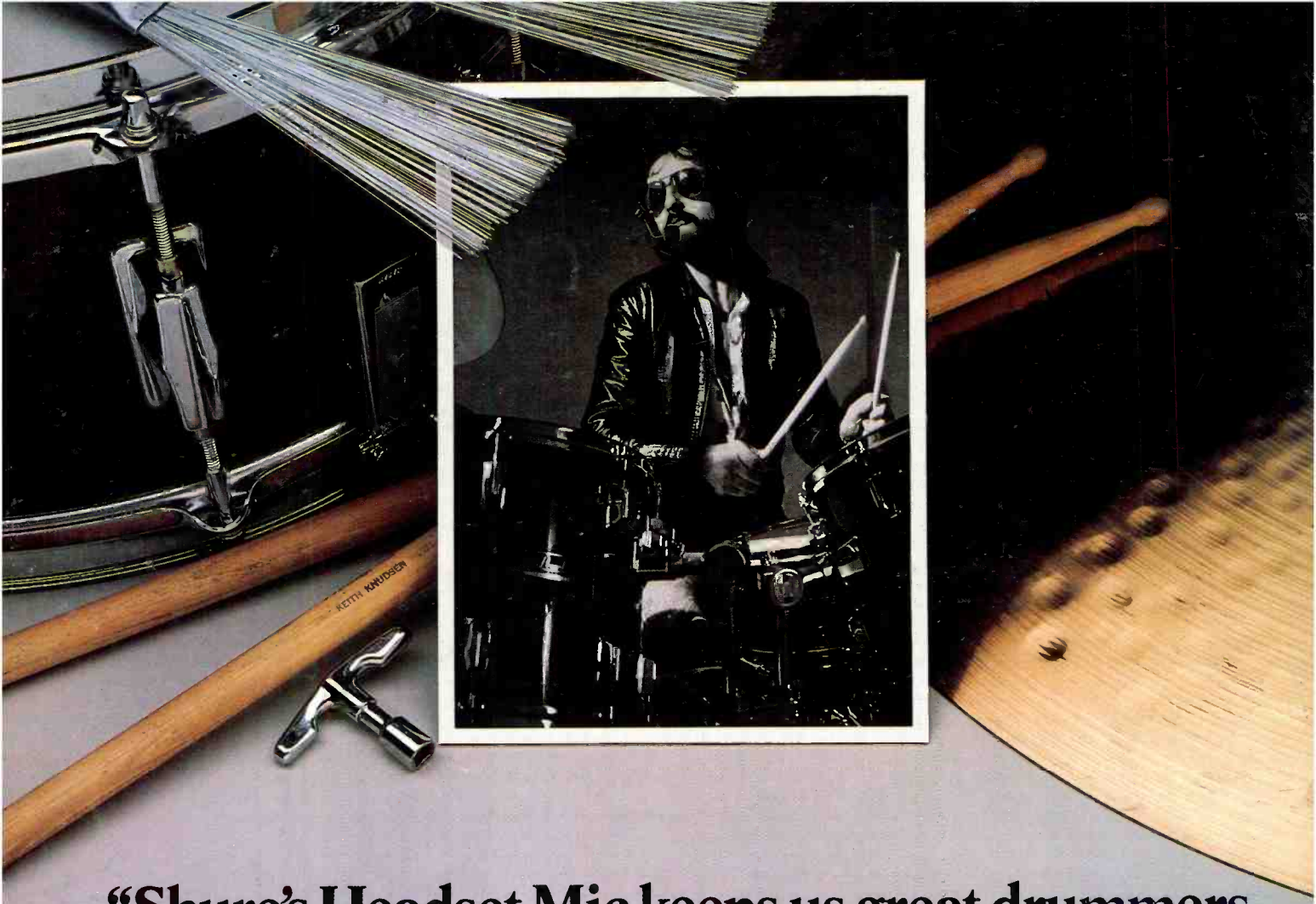

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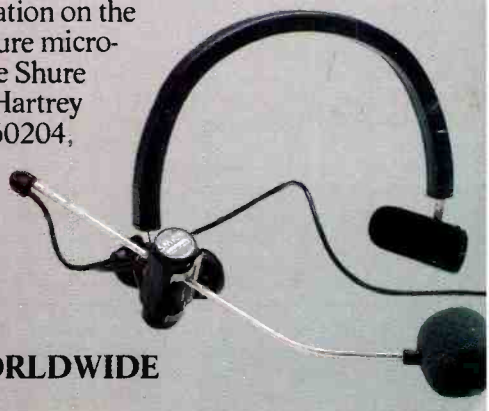

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