RECORDING ENGINEER/PRODUCER

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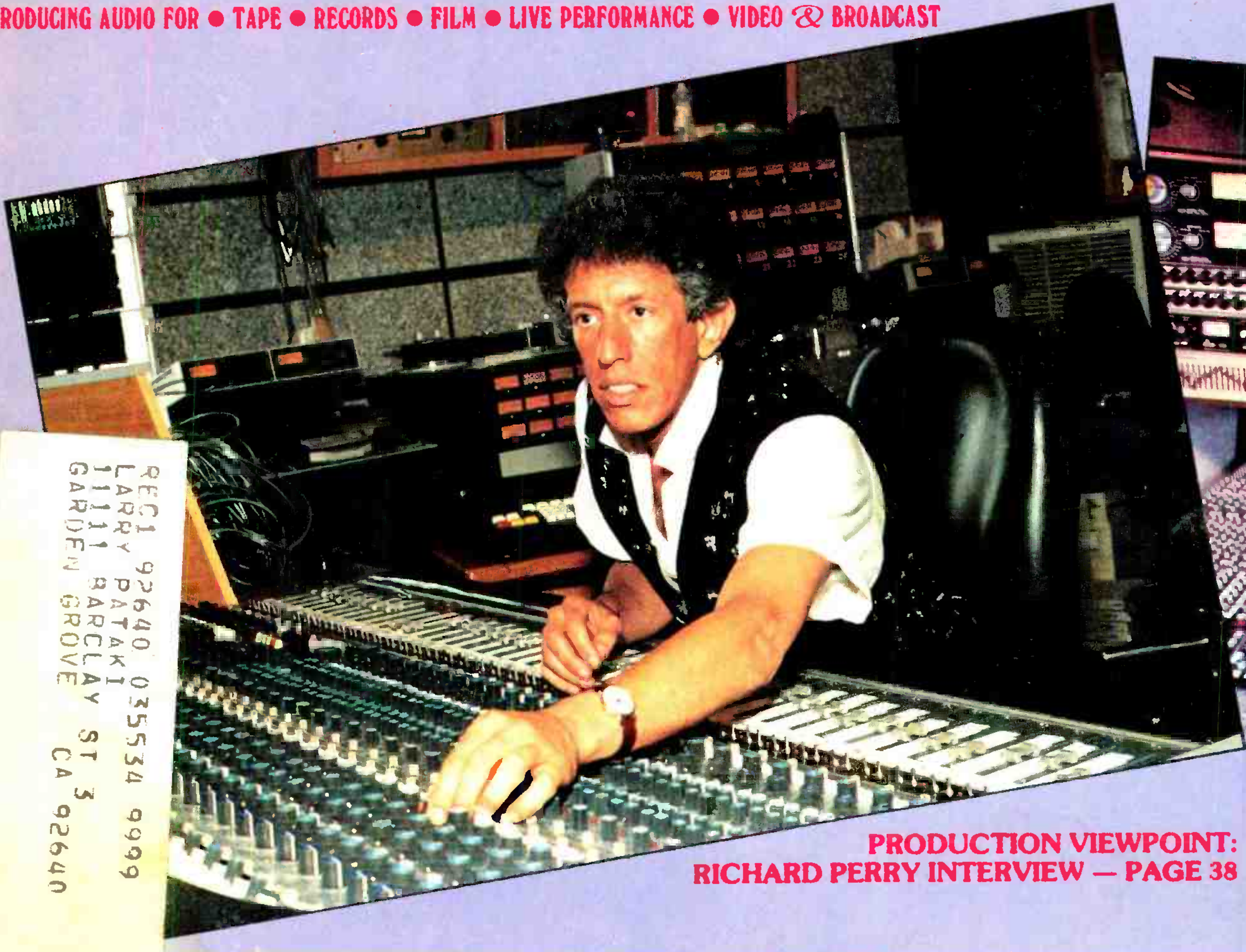
October 1984
Volume 15 — Number 5

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



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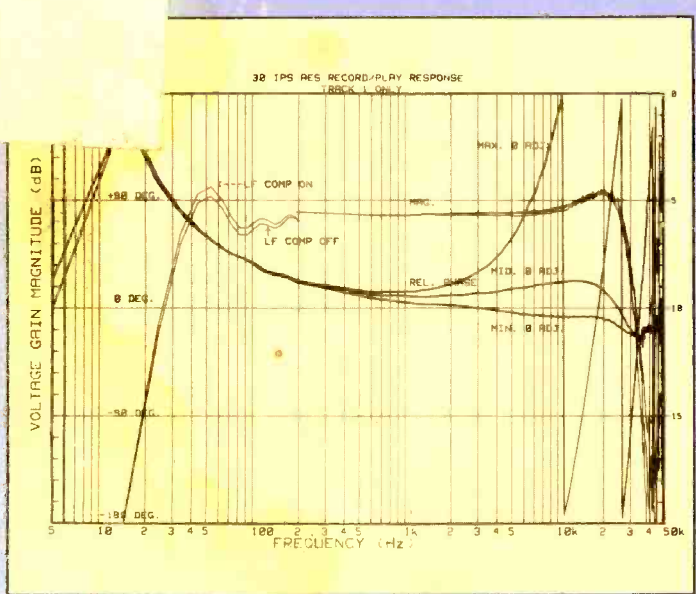

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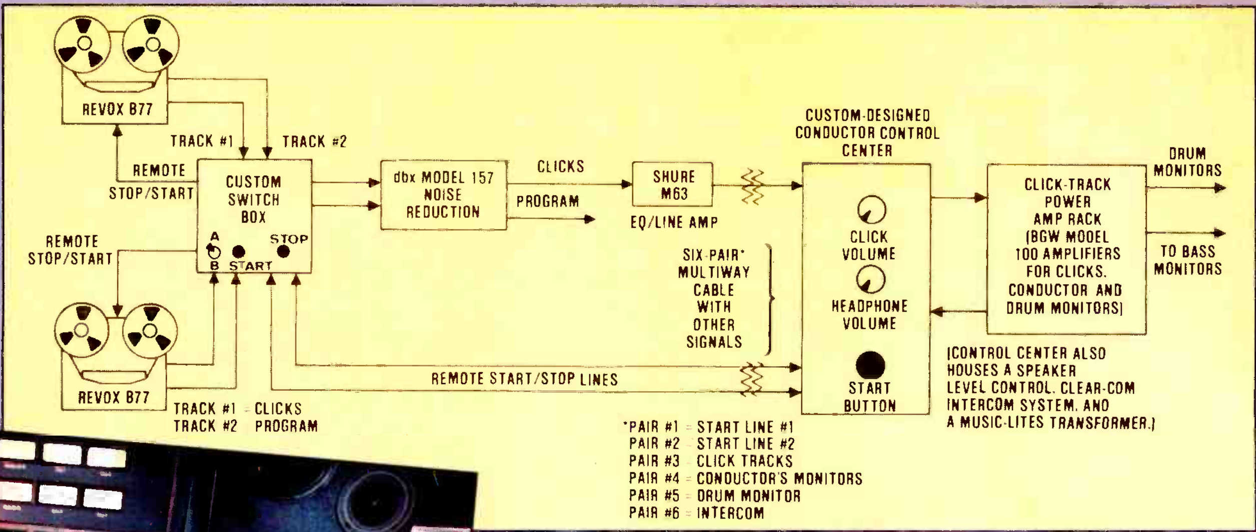

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— the magazine produced to relate recording **ART** . . . to recording **SCIENCE** . . . to recording **EQUIPMENT**.

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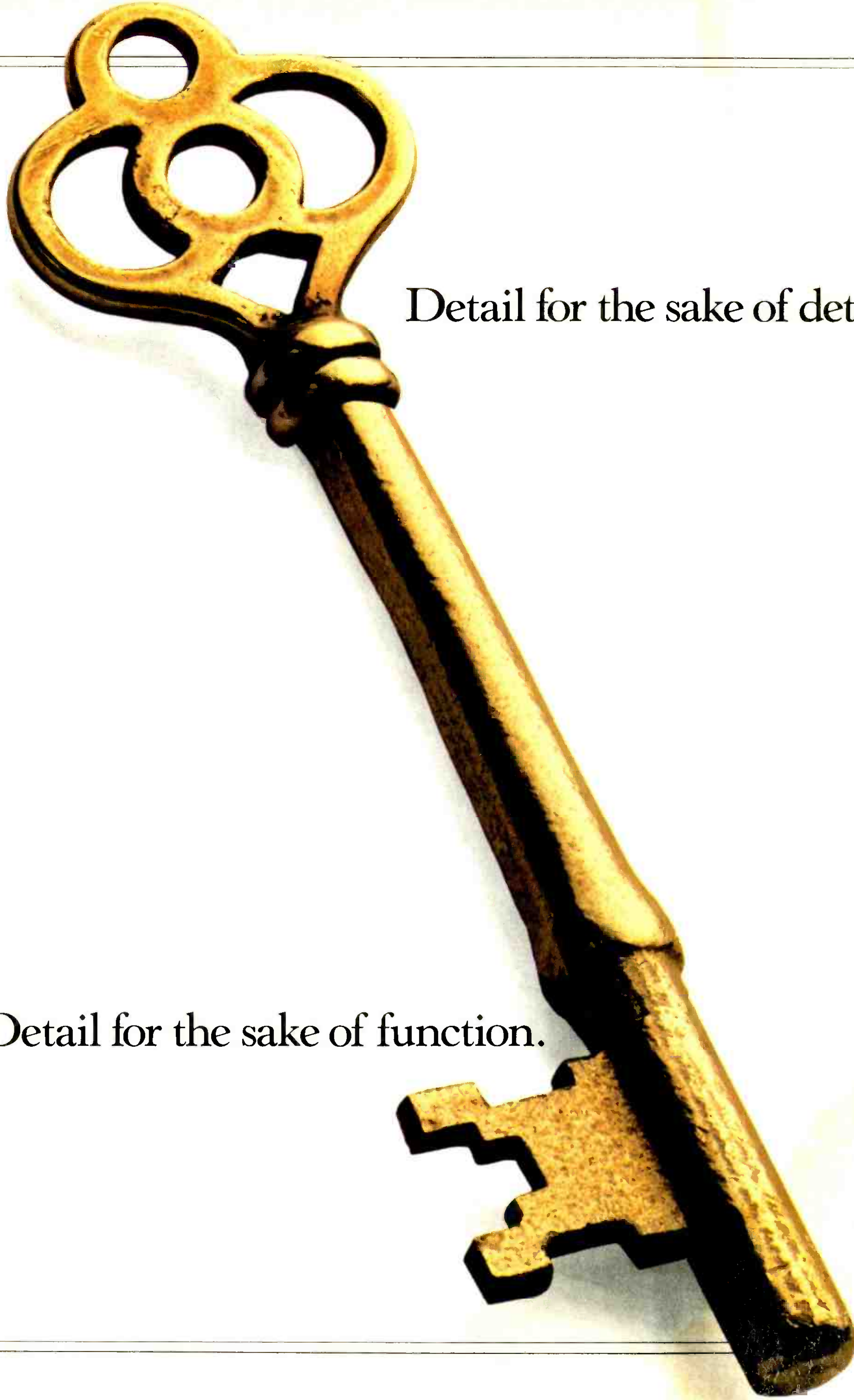

Detail for the sake of detail.
Detail for the sake of function.

Sound Workshop's attention to detail always produces the most sensible approach.

The design of a professional audio console encompasses a vast array of design decisions. It is Sound Workshop's relentless attention to detail throughout every critical design area that enables our consoles to make total sense.

Metering: This attention to detail has produced the most sensible high-resolution metering system available today. Why bother, when sonic excellence is the most important attribute of an audio con-

Conventional mechanical meters also available

sole? Because the metering system is effectively your "window" into the performance of the console signal path and plays a vital role in making the recording process more accessible and efficient.

Consider what the Series 34 has to offer in terms of metering:

- Forty segments for high resolution and a wide 40 dB dynamic range
- Three color display for the best response to peripheral vision
- Peak and Average modes with no gain scale shifting to permit accurate visual analysis of peak signal content
- One quarter dB resolution at "O Vu" for accurate calibration and tape machine alignment
- Variable intensity function allows LEDs to be adjusted for optimum brightness in a wide range of control room lighting conditions
- Individual calibration for Peak and Average modes
- High-slew balanced input circuit prevents ground loops and high frequency inaccuracies
- Modular design for easy maintenance access

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

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

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

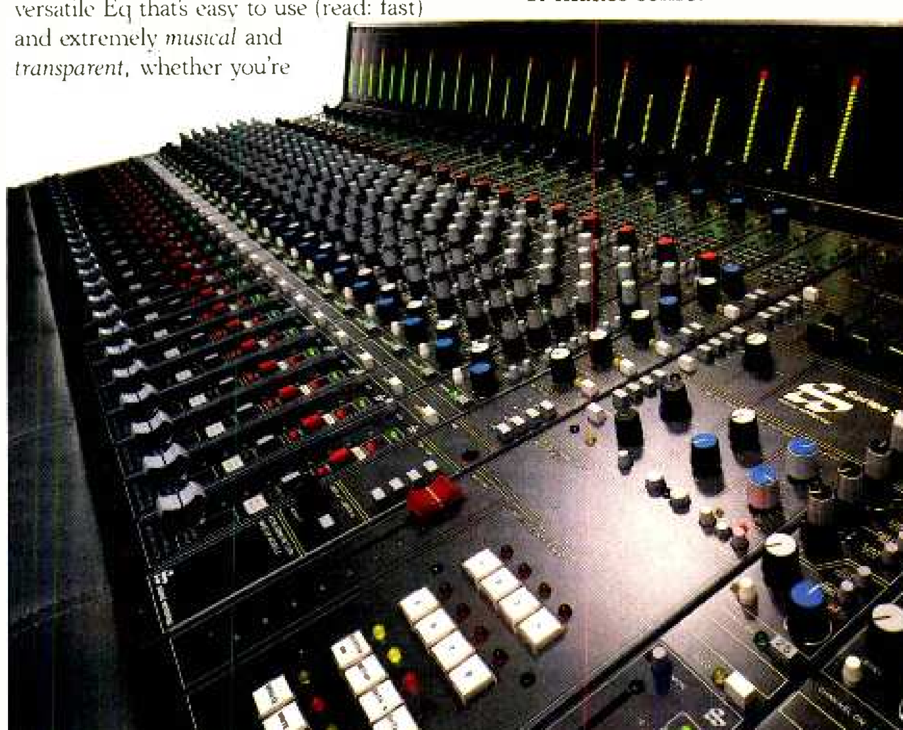
Equalizer combines continuous control and detents for reset-ability

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

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

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

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



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sound workshop

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AMPLIFIER PATENT

from: **Deane Jensen, president**
Jensen Transformer Inc.
North Hollywood, CA

Reluctant as I am to become engaged in a controversy with as honorable a gentleman and good customer as Welton Jetton, I have decided that I must respond to his recent "Letter to the Editor" [August issue of *R-e/p*, page 12] regarding his patent No. 4453131 Transformer Coupled Amplifier Circuit.

Indeed, Jeffrey Paullus has developed a superb transformer output circuit for the Auditronics product line which yields "transformerless specifications while retaining the benefits of a transformer." And, of course, I am enthusiastically in favor of applications which make best use of transformers; especially since these Auditronics products have used Jensen Transformers for over eight years.

Soon after Mr. Jetton's letter was published, I received some phone calls from customers who are planning to develop products using feedback around transformers. They expressed sincere concern about continuing their work without "stepping on anyone's toes," and I sensed that Mr. Jetton's announcement had an intimidating effect, as perhaps was part of its purpose.

Wondering how anyone could possibly have patented the idea of feedback around a transformer, since I knew that Bill Putnam's 1108 FET amplifier [shown here] and 1176 FET limiter of the 1960s had utilized transformer feedback with a tertiary winding and separate feedback paths for DC and very high frequencies, and also MacIntosh years before that, I researched the new Auditronics patent to find that its scope is, in fact, quite narrow.

Unlike the Sunset Sound circuit, which Don Wolford published in *R-e/p*, the Auditronics circuit uses an *inverting servo amplifier* for the DC feedback connected to the *non-inverting (+) input* of the audio op-amp. The Sunset Sound circuit uses a *non-inverting servo amplifier* for DC feedback connected to the *inverting (-) input* of the audio op-amp. Mr. Jetton stated in his letter that his patented circuit is "essentially identical" to the Sunset Sound circuit. Actually, it is not.

Although the *Abstract* of the patent reads:

"A transformer coupled amplifier circuit that eliminates transformer induced non-linearities by way of a dual feedback path without sacrificing any of the inherent attributes of general transformer coupled amplifiers,"

the *Claims* of the Auditronics patent are specifically limited to the *inverting servo amplifier* case:

"1. A transformer coupled amplifier circuit . . . said circuit comprising:
(a) input conditioning means . . .
(b) differential amplifier means . . .
(c) current booster means . . .
(d) output transformer means . . .
(e) alternating current feedback means . . .
(f) *inverting* amplifier [emphasis added] means for monitoring direct current voltage of said output signal of said differential amplifier means and for providing a direct current error signal, and
(g) direct current feedback means . . .

The patent is infringed *only* if the circuit uses *all* of the items in the *Claim* of the patent. You are free to use any circuit which avoids the use of the inverting servo amplifier for the DC feedback. The circuit topology used by Sunset Sound *does not infringe* on the patent.

We (Eric Benton, Don Wolford, Steve Hogan, and Deane Jensen) had investigated both the non-inverting and the inverting approaches thoroughly, and rejected the inverting servo amplifier method because it has difficulty with

. . . *continued on page 15* —

CIRCUIT DIAGRAM OF UNIVERSAL AUDIO 1108 AMPLIFIER

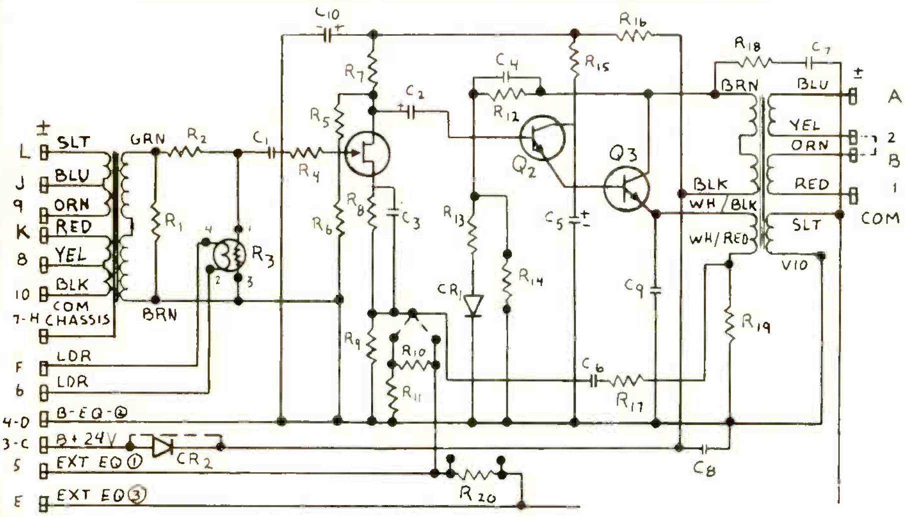
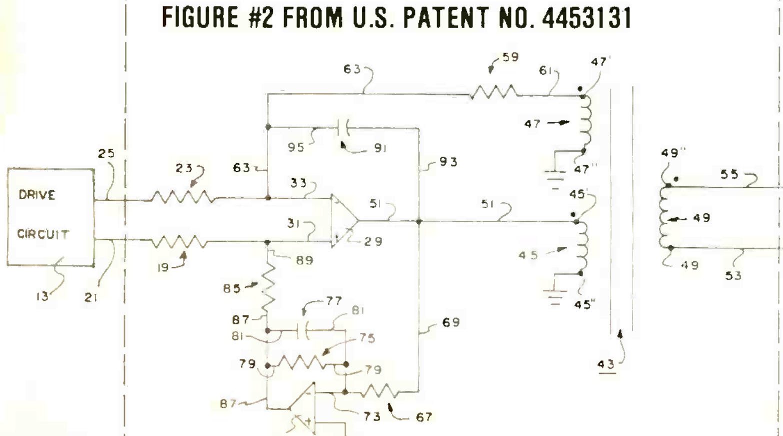


FIGURE #2 FROM U.S. PATENT NO. 4453131



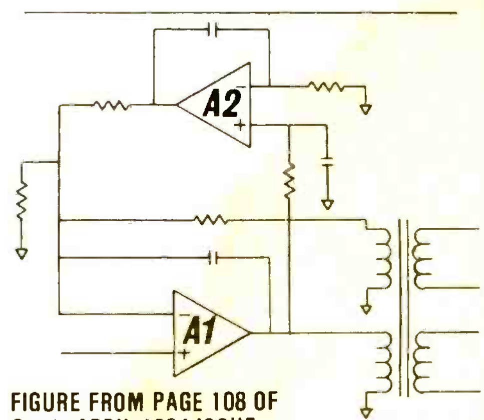
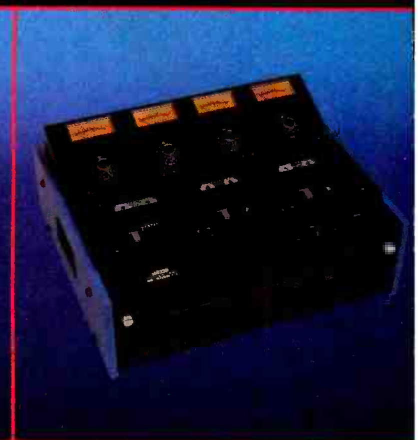


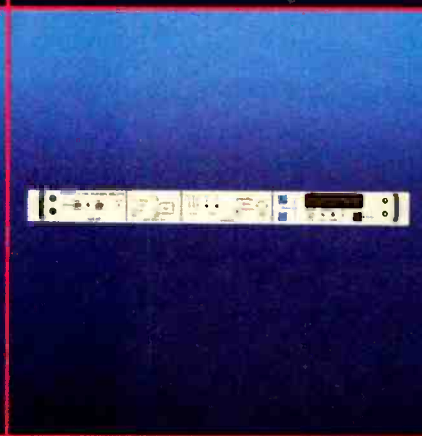
FIGURE FROM PAGE 108 OF *R-e/p* APRIL 1984 ISSUE

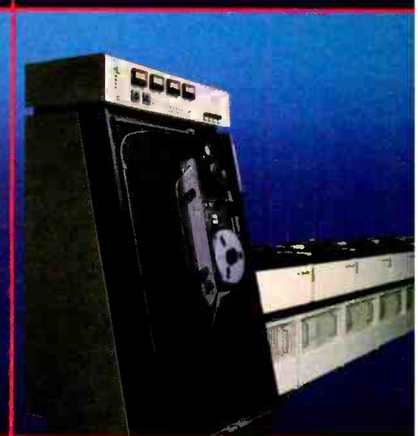
***The DP-4050-OM** is an open reel master reproducer, capable of driving up to 28 cassette slave units at 8:1 speed. The OM is fully automatic, with rewind-to-cue and repeat functions, and is available in versions providing 3.75 and 7.5 ips, or 7.5 and 15 ips.*

***The MARK III/4**, an affordable ½" 4-channel recorder for professional broadcast and audio post-production. It compares, feature-for-feature and spec-for-spec, with many more expensive ½" 4-channel recorders. And for top quality audio-visual programs, the BOII (a ¼" version of the MARK III/4) is the world's best ¼" 4-channel recorder.*

***The DP-4050-C2** cassette-to-cassette duplicator with two slave units, copies cassettes at 8:1 speed, duplicating both sides simultaneously in one pass, providing full stereo duplication. The C2 can be combined with additional slave units to reproduce up to 11 copies per pass, and will process a C-60 in under 4 minutes.*







***The MARK III/2** tape recorder delivers high performance at a price that will surprise you. It excels as a broadcast editing machine, or in studio mix-down and copy applications. The MARK III/2 features a single interface connector to SMPTE time-code-based editors, machine controllers or synchronizers.*

***The EC-400 Series** options for pilot tone resolve applications, and the EC-100 Series "in-machine" chase synchronizer modules, are designed to optimize the unique high performance capabilities of Otari tape transports. These options are another example of Otari's on-going product development program designed to keep your audio systems ready for the future.*

***The Otari DP-80** is the only 64:1 audio tape duplication system that is capable of running a 7.5 ips master tape. The system can be configured with from 1 to 20 slave units, producing up to 2880 C-45 cassettes per hour.*

Letters

— *continued from page 12 . . .*

sensitivity to external source resistance at the non-inverting input of the audio op-amp.

According to Jeff Paullus' phone conversations with me, and also his wording in the patent, he believes that the idea of feeding the DC feedback into the other input isolates it avoiding "swamping out" the effect of the transformer feedback; because now the DC feedback resistor does not appear to be connected across the transformer feedback resistor. Actually, it is only the frequency selective characteristics of the servo integrator lowpass filter and the transformer highpass filter effect that cause isolation between these two feedback paths.

Eric Benton, of Sunset Sound, used the *Optimization* feature of our COMTRAN program on the Hewlett/Packard computer, to calculate the integrator response to match (complement) the transformer response below one Hertz to avoid a peak or notch at low frequencies.

We have also found that the real key, to making these DC servo circuits work, is the addition of *input bias injection* as shown in the *twin servo microphone pre-amplifier* of the Jensen Transformers JE-16-A/B transformer data sheet.

And Steve Hogan has used the computer to derive a circuit for avoiding loss of low-frequency feedback during transformer saturation, which is most probably patentable, but we would rather publish it for the benefit of all.

See the next issue of *R-e/p* for the complete *Public Domain circuit* for *DC Servo Amplifiers with and without Transformers.* □□□

Reply from: Welton H. Jetton, president, Auditronics Inc.

Having reviewed Mr. Jensen's reply to my "Amplifier Patent" letter published in *R-e/p*, August, 1984 edition, I feel that further clarification is needed.

It was certainly not my intent to offend, embarrass or intimidate Don Wolford and his associates at Sunset Sound. Having spent more than 25 years designing and building audio consoles, I can fully appreciate what they have accomplished. I congratulate them for their advance of the art.

Auditronics has a duly granted patent covering circuitry similar to, but not identical, to that used in the output stages of the Sunset Sound console. However, the point I want to make is this: Those who may contemplate modification of Mr. Wolford's or Mr. Jensen's "Public Domain" circuit could possibly infringe on the Auditronics patent. Those considering such an approach should obtain a copy so infringement may be avoided.

I am not a patent expert, but I know that cases of this type can only be resolved by the courts. We (Auditronics) cannot say with absolute certainty that a circuit infringes on our patent. We can say, however, that we feel there could be a problem of possible infringement and let the people who decide these things pass judgement. I would like to quote the introductory paragraph to the claims section of the patent:

"Although the invention has been described and illustrated with respect to a preferred embodiment thereof and a preferred use thereof, it is not to be so limited since changes and modifications can be made therein with or within the full intended scope of the invention."

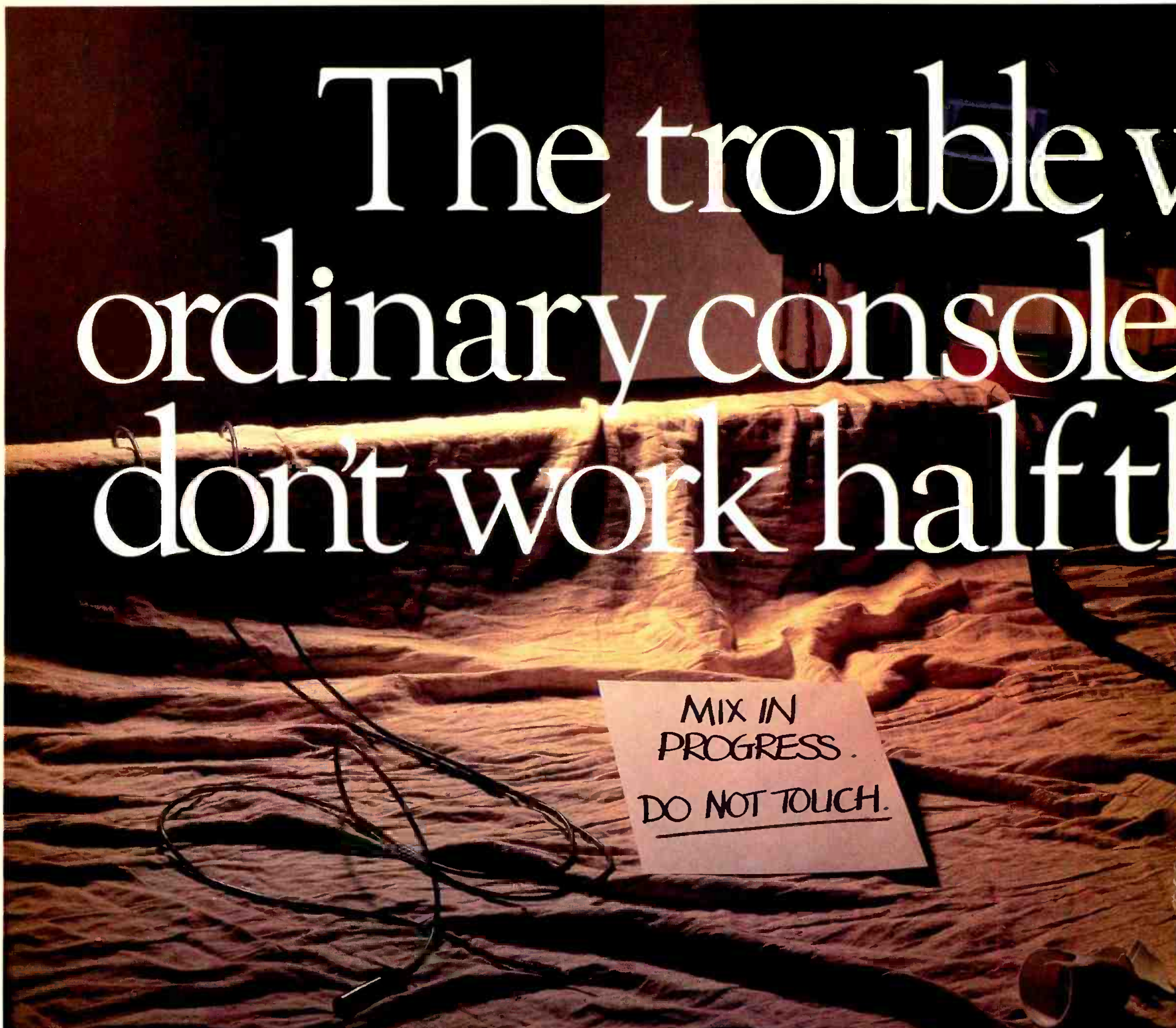
Additional comments by Jeff Paullus, inventor:

As it turns out, the last several paragraphs of Mr. Jensen's letter cause *me* more concern than the actual issue of infringement.

To state that the "inverting servo" has "difficulty with sensitivity to external source resistance at the non-inverting input" is misleading. This presumes that the circuit is being fed at the non-inverting input. Also, this effect is presumably at below 1 Hz. We decided that flat low-frequency response to 5 kHz (-1 dB) would suffice in the real world. The passband gain error caused

. . . *continued on page 20* —

The trouble w
ordinary console
don't work half th



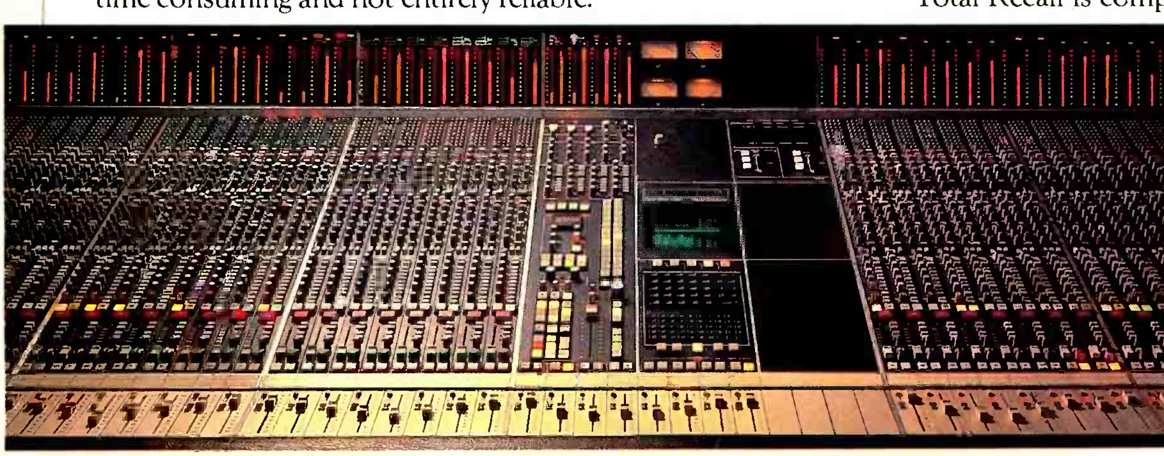
It's a situation that every studio manager recognises. A client has been in, done some work, and departed to return some time later. Expecting to find the desk as it was left.

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

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

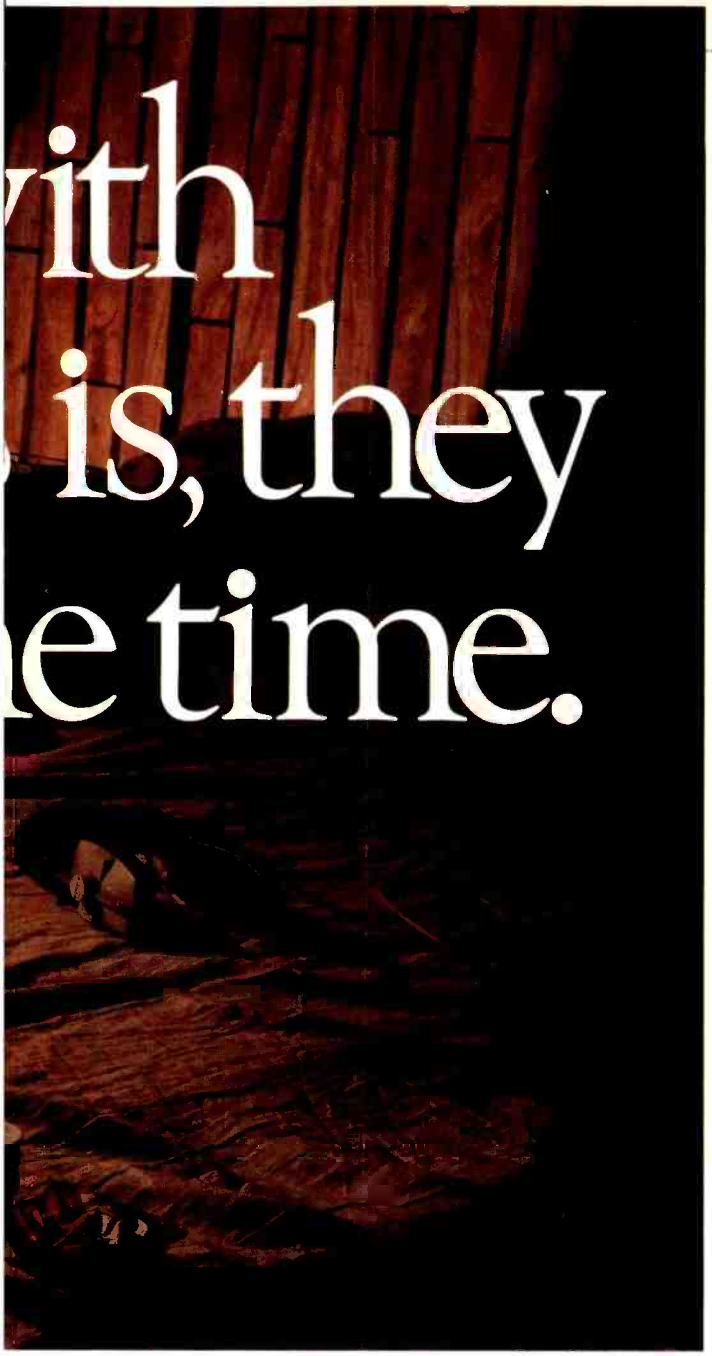
At Solid State Logic, however, we've developed a rather more practical solution to this dilemma. We call i the Total Recall System.

Total Recall is completely independent of all audi paths and allows the consol settings to be permanently stored on floppy discs withi a few seconds.



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

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



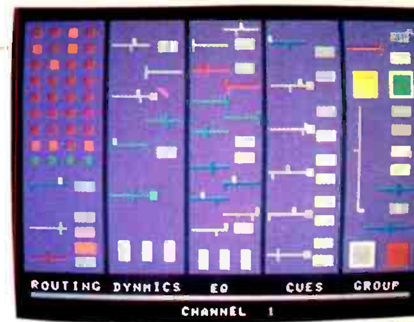
to save time at a later re-mix. And engineers can even store their personal EQ and dynamics settings and create their own libraries on floppy disc.

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.

Name ____________________

Position ____________________

Company ____________________

Address ____________________

Solid State Logic, Churchfields, Stonesfield, Oxford OX7 2PQ, England. 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.

Letters

— continued from page 16 . . .

by high source impedance is usually of more concern, and optimizing frequency response below 1 Hz is of academic interest only.

As for "input bias injection" being "the key to making these DC servo circuits work;" nulling inherent amplifier offsets will *always* aid a servo circuit striving for DC accuracy, but until circuit gain becomes such that device dependent offsets become excessive, the benefits are minimal when weighed against the extra time (expense) of trimming.

After reading the description of "Circuit action as interpreted by Deane Jensen," I can only suggest that he read the "Prior Art" and "Abstract" portions of the text to find out my intended meaning of "Swamping out." I assure you that I understand integrator circuit operation. □□□

Reply from: Deane Jensen

"The *Claims* of the Auditronics patent are specifically limited to the *inverting servo amplifier* case . . . You are free to use any circuit which avoids the use of the inverting servo amplifier for the DC feedback. The circuit topology used by Sunset Sound *does not infringe* on the patent."

Excuse me for repeating myself, but it is evidently necessary.

"Those who may contemplate modification of Mr. Wolford's or Mr. Jensen's 'Public Domain' circuit could possibly infringe on the Auditronics' patent" *only if the modified circuit uses an inverting amplifier for DC feedback.*

Anyone who can read plain English can determine the above statements from the wording in the *Claim* of the patent.

Mr. Jetton warns us against using his specialized treatment of transformer feedback with an inverting amplifier for the DC feedback, stating that modifications of the non-inverting Sunset Sound circuit might infringe his patent, with words of legal overtone.

I also recommend against the use of his circuit for reasons based on engineering principles.

The patented circuit has no advantages over other feedback methods in common use, as shown in the Sunset Sound circuit. As always with patents, the inverting circuit was patentable because no one had ever done it that way before; because anyone with a thorough understanding of feedback would not have chosen that method.

No one will ever need to use the patented circuit, because there are better ways to accomplish the same result with less sensitivity to external circuit values.

Conclusively, I predict that no one will ever license the patented circuit.

The tone of Mr. Jetton's first "Letter to the Editor" is inappropriate for a narrow patent, and I object to his attempts to imply that the scope of his claim is so broad as to include the Sunset Sound circuit. It clearly is not.

Let us not be subjected to any more unfounded attempts of intimidation regarding the use of public domain circuits. ■■■

EXPOSING AUDIO MYTHOLOGY

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

by John Roberts

I hope I don't bore you with all these digital pieces (bits?) in my regular column, but as with any new ball game, there are new rules to learn.

In part one of this month's offering I'll be discussing the repeatability of digital recordings made with magnetic tape. In part two, we'll take a look at DC Servo amplifiers which, contrary to popular belief, usually do have capacitors in their signal paths.

Digital is Forever . . . Well, Almost Forever

In direct A/B comparisons, the best analog tape machines can give digital recorders a run for the money, with some listeners preferring the "sound" of analog. However, if we repeat this comparison a week later, the analog recording will have lost some "sparkle." High frequencies recorded onto magnetic tape tend to relax or self-erase over time, while the digital recording will sound identical to its first playing, and will now be clearly superior.

This robustness of digital recording is one of its most desirable characteristics. By definition, a recording should sound the same every time it is played back. Digital recordings made on magnetic tape also suffer from this same relaxation of high frequencies; however, the nature of storage as binary bits and additional digital-domain processing all but eliminate the audible effects of such deterioration.

Redundancy, Interleaving, Error Correction, Interpolation and, when all else fails, Blanking.

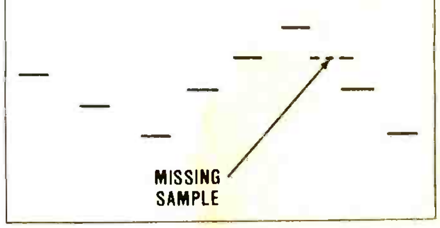
I recall the rather difficult time I had one evening trying to convince a friend that digital playback would be identical every time. He had a great deal of experience with analog recording, and questioned how all those digital ones and zeros would cope with the vagaries of analog tape. He was quite correct in his perception that the digital signal would get trashed by the occasional dropout and the like. Since PCM digital is most sensitive to losing even a single bit, there should be a problem. This was well considered by the digital designers, however, and several different techniques are used to deliver mostly error-free playback.

The obvious compromise in magnetic recording is packing density. An audio signal sampled at 44.1 kHz and digitized to 16 bits will generate a data stream of over 700,000 bits per second. To magnetically record a number of such signals with acceptably low bit error rates (like none), would require prohibitive amounts of tape because of the high tape speed required, and it would still be impossible to promise no errors. The design solution was to plan on there being errors. By allowing a relatively few random errors to occur, high packing densities could be achieved. By adding a few extra bits, and using an error-correcting scheme, simple single-bit errors could be completely corrected, allowing perfect playback.

In cases where too many bits are lost to recover the original word, the digital processing will use "interpolation" (as shown in Figure 1) to substitute a replacement word that is exactly halfway between the amplitude of the preceding and following words. Generally, this estimate will be very close to the actual value, and typically not audible. There is a catch, however. For such error-correcting schemes to work, the errors must be random (not consecutive). Magnetic recording tape, because of physical imperfections, tends to have dropouts that last too long for either error correction or interpolation to save.

To deal with these longer errors, a third technique called "Interleaving" was developed (see Figure 2). By rearranging the digital words into nonconsecutive order when recording onto the tape, and then re-ordering for playback, dropouts that are several words long will appear as several one-word

Figure 1: Interpolation of Error.



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dropouts dispersed through the samples. As one-word errors can be interpolated or, if enough redundancy is built in, replaced, even the dreaded dropout can be effectively handled.

As long as the errors are within the range that can be corrected by the error correction, the playback will be perfect (or as perfect as it gets). Error rates that cause occasional random interpolations will not sound bad but, as the frequency of interpolations increase, the sound quality will be affected. At some point the processor will give up trying, and just blank out the signal.

It is interesting to consider that head alignment on a digital machine can effect the error rate when replaying tapes made on a different machine. Any error rate beyond what can be completely corrected will make a sonic difference, however slight.

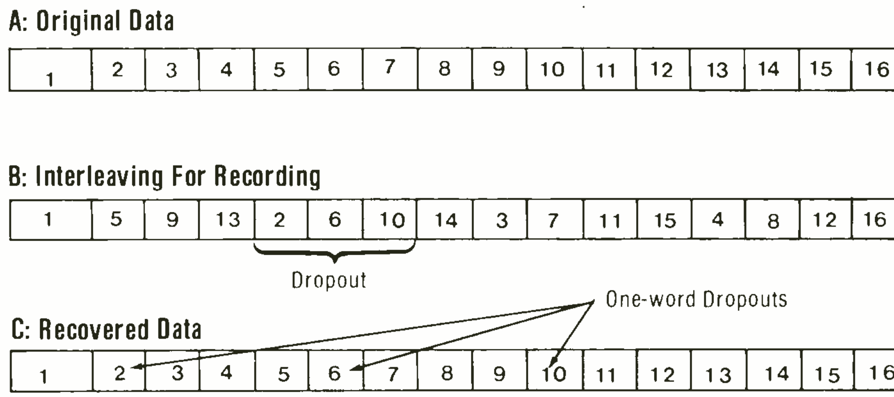


Figure 2: Interleaving Scheme for Error Concealment.

So What Does it All Mean?

No doubt the designer has included enough redundancy and error correction to deliver perfect or near perfect playback over the reasonable life of the master. Furthermore, the digital tape need not be particularly linear, nor handle a wide range of frequencies, allowing the tape designer to optimize for the digital signals. However, the digital signal is mostly high-frequency energy, and likely to start erasing itself the minute it is recorded. For this reason, it seems there would be some merit in periodically dubbing masters via a digital-to-digital process to freshen up the recordings. (You would want to avoid conversions into and out of the analog domain as that might hurt more than it helped, but running through the error correction would fix any simple errors caused by relaxation.)

I regret that I cannot offer any firm suggestions regarding how often tapes should be refreshed, and I expect newer tape formulations may be better than early ones. I would be glad to report feedback from tape companies, machine manufacturers or studio users that can offer more specific guidelines on the care and feeding of digital masters; they should indeed last forever if properly cared for.

DC SERVO AMPLIFIERS

DC Servo amplifiers have become popular in some circles, because of the perception that they eliminate capacitors from the signal path. Figure 3 shows a block diagram of a typical servo amplifier circuit. At first glance it appears that the signal is indeed unmolested by those nasty capacitors but, alas, there is no good way to create a LPF (low pass filter) function without them. An RL circuit would probably be worse than the RC it replaced, and a digital LPF would be much more expensive and well . . . digital.

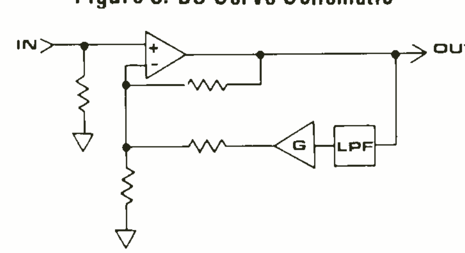
As a result, there will be capacitors in this parallel (servo) feedback path, and any artifacts of non-ideal capacitor performance will be re-introduced at the input to mix with the signal.

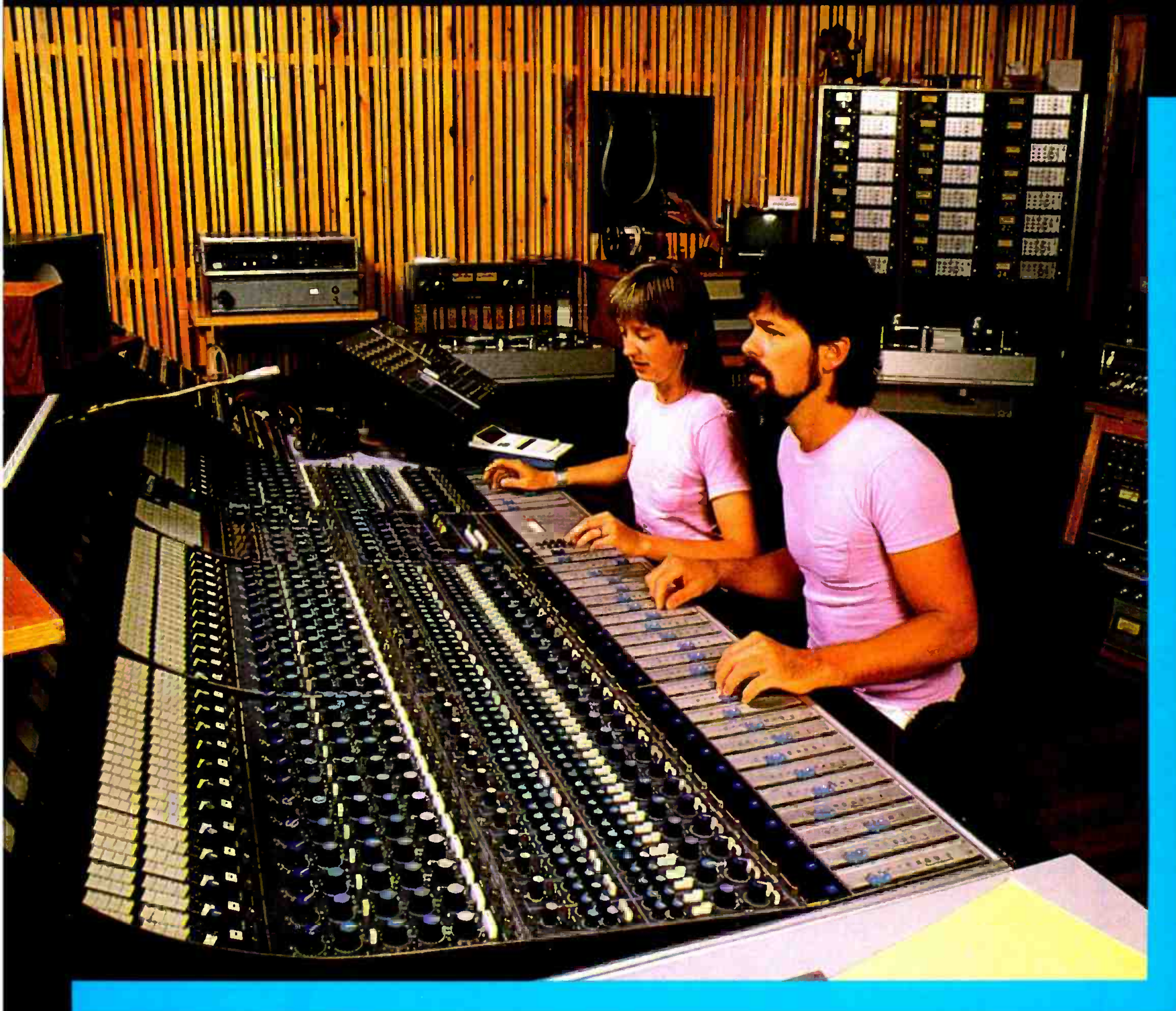
If Servos Don't Eliminate Caps, Then What Are They Good For?

By these revelations I don't suggest that servo designers are somehow misinformed. Let's take a look at what servos do right. You may recall from my February 1984 column that large-value capacitors (usually electrolytic) do not perform very well when operated into low impedances. To capacitor-couple into a 600-ohm impedance with 20 Hz (-3 dB) response requires almost 15 microFarad; to couple a 200-ohm microphone would need several times that. Such high-value capacitors are large, and not very ideal performers.

The servo topology allows a simple circuit to be DC-coupled at the input and output, while still not passing direct

Figure 3: DC Servo Schematic





current. The op-amp (buffer/gain stage) in the servo feedback path allows the LPF function to be generated with high-impedance resistors and small-value capacitors. Since small film capacitors are relatively inexpensive, and near-ideal performers, these components will not degrade the sound. (Note: If a servo was built up with lousy capacitors and, for that matter a low-performance op-amp, it could sound just as bad as putting those components in the direct signal path.)

In addition to the ability to inexpensively utilize better components, the servo designer can also reduce the AC contribution of the servo path by varying the feedback ratios. However, this approach will limit the ability of the servo to block DC from input to output.

Which leads to just about the only thing I can think of that's wrong with servos. When they fail, servos almost always will fail with some significant amount of DC on the output. Likewise, some designs will not be capable of blocking the DC from a failed earlier stage. A true capacitor-coupled output will almost always protect against its or an earlier stage's failure.

I expect servo designs will grow in popularity, as the price of high quality op-amps continue to decline. It is already cheaper (and smaller) to use a bi-FET op-amp with some small film capacitors in place of a 15 microFarad film output coupling capacitor. The undesirable failure mode will only be a problem if the following stage is incapable of blocking DC, or doesn't offer some other form of protection, such as power-amp shutdown. ■■■

VISUAL MUSIC SCENE

Adapting Record Session Techniques to Film Sound Re-recording:

Music Mixer Neil Brody Working on *PurpleRain*

by Adrian Zarin

The influence of Music Video on feature-film releases continues to grow, as filmmakers devote increasing attention to the all-important 14- to 20-year-old segment of the film-going public. Summer hit movies like *Ghostbusters,* and Prince's *Purple Rain* demonstrate the important and diverse roles rock music can play in feature films. And let's not forget that the synergistic, mutually supportive sales clout of records, feature films and video clips stands as a virtual guarantee that the rock-oriented feature film is a genre that's likely to be around for some time to come.

It is a genre that also calls for a special breed of music mixer. The job demands all the skills of a record-session engineer to bring the sound of contemporary records into the movie theater. At the same time, it calls for a mixer that can work with the specialized technology and procedures of the film world — a world that, in many respects, is markedly different from that of records.

Representative of this emerging "new breed" of mixers is Neil Brody, who served as music mixer for *Purple Rain.* After 12 years of engineering records for such artists as Neil Diamond, Robbie Robertson and The Band, and Sammy Hagar, Brody crossed over to films some two years ago, amassing a list of credits that includes *Frances, Best Friends, Dreamscape, The Last Waltz, Suburbia, Running Brave,* and *Children of the Corn,* among others. Currently a staff mixer at Hollywood's Ryder Sound Services, Inc., Brody's skills were loaned to Glen Glenn Sound, where the film soundtrack for *Purple Rain* was re-recorded.

According to the engineer, it was Brody's combination of record and film experience that led to his selection as music mixer for Prince's cinematic tour-de-force. "For *Purple Rain*, they wanted somebody who understood both sides of the street: they wanted a record sound, but within the context of film sensibilities. It called for an ability to draw on both film and record techniques without leaning too heavily on either. We went for a real rock sound. The director [Albert Magnoli] wanted music that was as forceful as we could make it — when forcefulness was called for — and as subtle as we could make it when subtlety was called for. The main idea was to keep it hot. They didn't want a typical film mix; they wanted a *hot* mix."

Brody is quick to point out that, while assignments of this nature call upon the film mixer to *parallel* the sound of contemporary records, the job involves much more than just reproducing recording studio techniques on the film dubbing stage.

"The fact is, you're working in two totally different environments," he comments. "When you are mixing a record, you are listening to two monitors that are fairly close to you. When you're doing a film mix, you're listening in a much larger space on speakers that are 30 feet away from you, and behind a screen. Plus, if you're talking about a Dolby Stereo release, you've got surround speakers and discrete center

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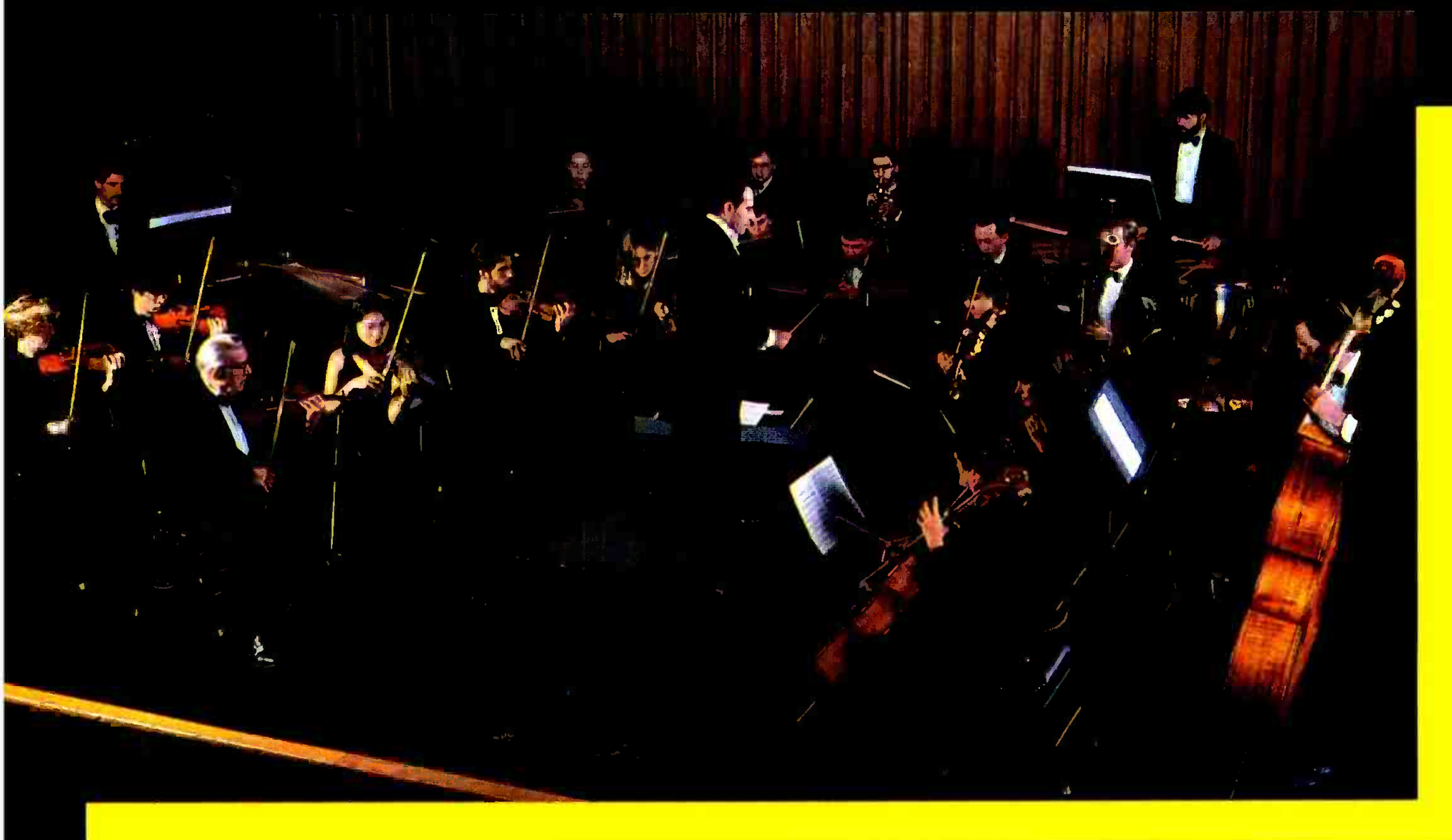

channel information. It's a very different experience. I had a lot of valuable help from Gary Bourgeois [dialog gaffer at Ryder] in making that transition."

Record-to-Film Transition

The record mixer coming into film has quite a bit of adjusting to do. One of the biggest adjustments, according to Brody, is acquiring a visual perspective in approaching a mix. "The main thing is that you have to interpret the emotion that's on the screen. More than effects, or even dialog, music leads the viewer to an emotional response to a scene. Music on a record is not interfacing with any other medium, so your only concern is to make it sound as big and as good as possible. On a film, you're always basing your mix on the picture, paying close attention to the footage counter and the cue sheet. Even with something like the concert scenes in *Purple Rain*, where the music is really a central consideration, the mixing is interpretative.

"To give an obvious example, on one shot you'll have Prince up on the stage; the music has to be big and loud. In the next shot, you see another group backstage, and the same music now has to sound as though it is coming from 200 feet away, through concrete walls. In this particular instance, we used extensive EQ and a Lexicon 224X [digital reverb] to create a believable effect. The 224X enabled us to dial up just the kind of echo we need for those and other scenes."

Throughout the *Purple Rain* mix, Brody worked closely with dialog mixer Jim Cook and sound effects mixer Bob Harman of Glen Glenn. (Glen Glenn effects mixer John Asman also became involved in mixing the audience tracks after the project had gotten underway.) The interactive nature of film audio work, according to Brody, is another area where the record engineer coming to the motion-picture world for the first time will have some major adjustments to make.

Compared with their film counterparts, record engineers enjoy quite a bit of autonomy. They do everything from miking the drums to razor-blading the tape, essentially answering only to the producer and, in some cases, the artist. Re-recording film sound involves a much larger number of people, all of whom play very strictly defined roles. In the audio domain alone, the music mixer will coordinate his efforts with the producer, director, music editor, composer of the score (on some occasions), and, most directly, the two companion mixers on the dub (handling dialog and effects).

"There is a truly symbiotic relationship between the three mixers," Brody explains. "All of us have to be aware of what the others are doing in every single frame. There's only so much sonic 'space' available. It's like a suitcase: you can only pack it so full. If you organize the way you pack the suitcase, you can get more in it than you could if you just

Re-recording engineer Neil Brody

threw things in. The music mixer will therefore carefully watch what frequency ranges the dialog or effects are in. If there is a lot of midrange in the sound effects, I might tend to pump a little more bottom into the music, and not push the midrange too hard. The main thing, of course, is to stay out of the way of the dialog.

"Things are in a constant state of flux in a film mix. You're a lot more active while mixing all over the board than you are for records. The music may be up really loud at one moment, and then, as soon as the dialog comes along, you'll have to really pull it way back. While some of these level changes are very extreme, all of them should be virtually imperceptible to the audience. It's up to the three mixers to make the transitions smoothly. Equalization works hand in hand with level changes in doing this. When you bring the music down very low under the dialog, you might push the bottom a little to give the feeling that the music is still there. All of this requires very careful coordination."

Accommodating the Dolby Stereo Matrix

For Brody, *Purple Rain* represented a special challenge — one that called upon all of his combined record and film mixing acumen. The master tape provided for dubbing to film was a two-track mix that Prince had made for the soundtrack album. Prince's mix, moreover, was relatively short of discrete left and right information, relying to a large degree on digital delay and other effects to create a sense of three-dimensional width. A departure from standard film practice (where three- and four-track masters are more commonly furnished for dubbing to picture), the supplied two-track music mix presented salient problems in two areas: the Dolby Stereo format, and equalization.

"Generally, in film work you're given a three-track master at least," Brody explains, "containing left, center and right information from which you then build your surround channel. In *Purple Rain*, we had only a phantom center, and had to build the surround channel from the left and right channels. At the same time, we didn't want it to sound like it was the same information coming from everywhere. The challenge was getting each channel to sound discrete while keeping intact the continuity of the mix that Prince had given us."

Brody's plan for accommodating Prince's two-track mix to the Dolby Stereo format was to allow the phantom center to provide center-channel information, rather than feeding a *discrete* center channel which, he felt, would detract from the stereo spread. In normal film practice, a four-track mix (left, center, right and surround) is prepared, and then encoded through the Dolby Stereo matrix to produce a Lt-Rt printing master, from which Dolby Stereo optical prints are made. These two tracks are then decoded at the individual venues to retrieve the original LCRS four-track mix.

For the center-channel information on *Purple Rain*, however, Brody relied on the information from the left, right and surround channels, some of which is normally fed back into the center channel as a routine part of the matrixing process.

"On its own, the matrix holds back some of the surround-channel information, and some of the left and right channel information, and sends it to the center," Brody elaborates. So, rather than feed the center, I just allowed the matrix to fold back what it was going to fold back into the center and tried to keep the wings [left and right channels] as wide as I could.

"For the surround channel, I wound up using a left-minus-right [L–R] format; taking the right channel, putting it out of phase with the left channel, adding the two together, and putting that in the rear channel [surround channel]. We found that putting the left and right channels out of phase gave us a more separate, discrete rear channel. That was the recommendation of David Gray, who, along with Max Bell, were the Dolby representatives for the project. We also added some digital delay to the surround channel at times, to make it more expansive and to create more of a discrete quality between the surround and the front channels. Without these measures, we would have ended up with something that sounded much more mono than it actually sounded in the end."

Another consideration was the effect of the matrix on the overall frequency content of the mix. An area of concern whenever the Dolby Stereo format is used, it became especially critical in *Purple Rain*, given the special equalization problems (described below) posed by the project.

"The matrix itself imposes certain restrictions," Brody notes. "There are things that you can and can't do. It's not like making a stereo record, where what you put on the tape is what you get back. The matrix can only handle a certain amount of level, and a certain amount of bottom end. You constantly have to keep these limits in mind when you work with it. For that reason, you monitor through the matrix even when you're

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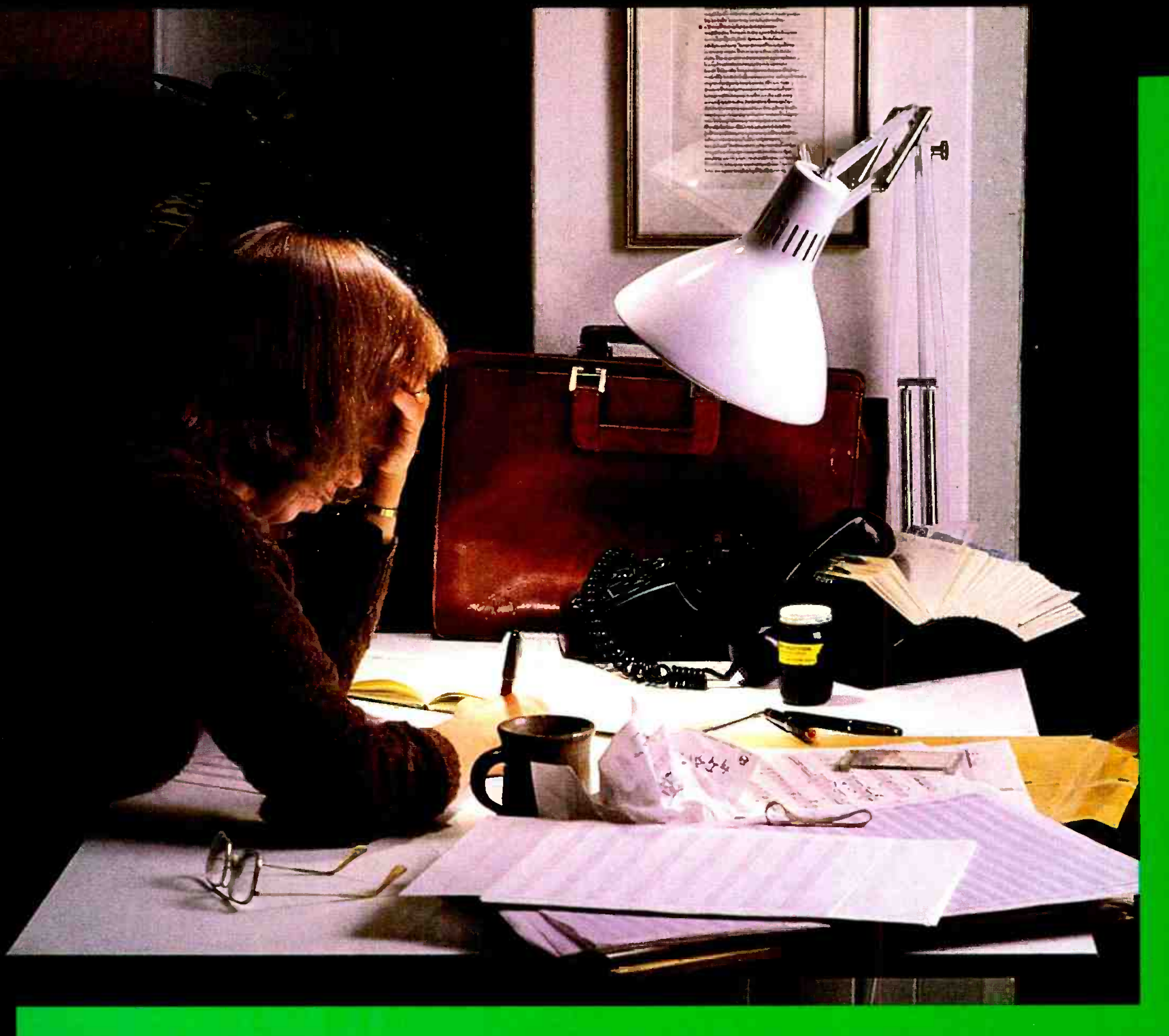

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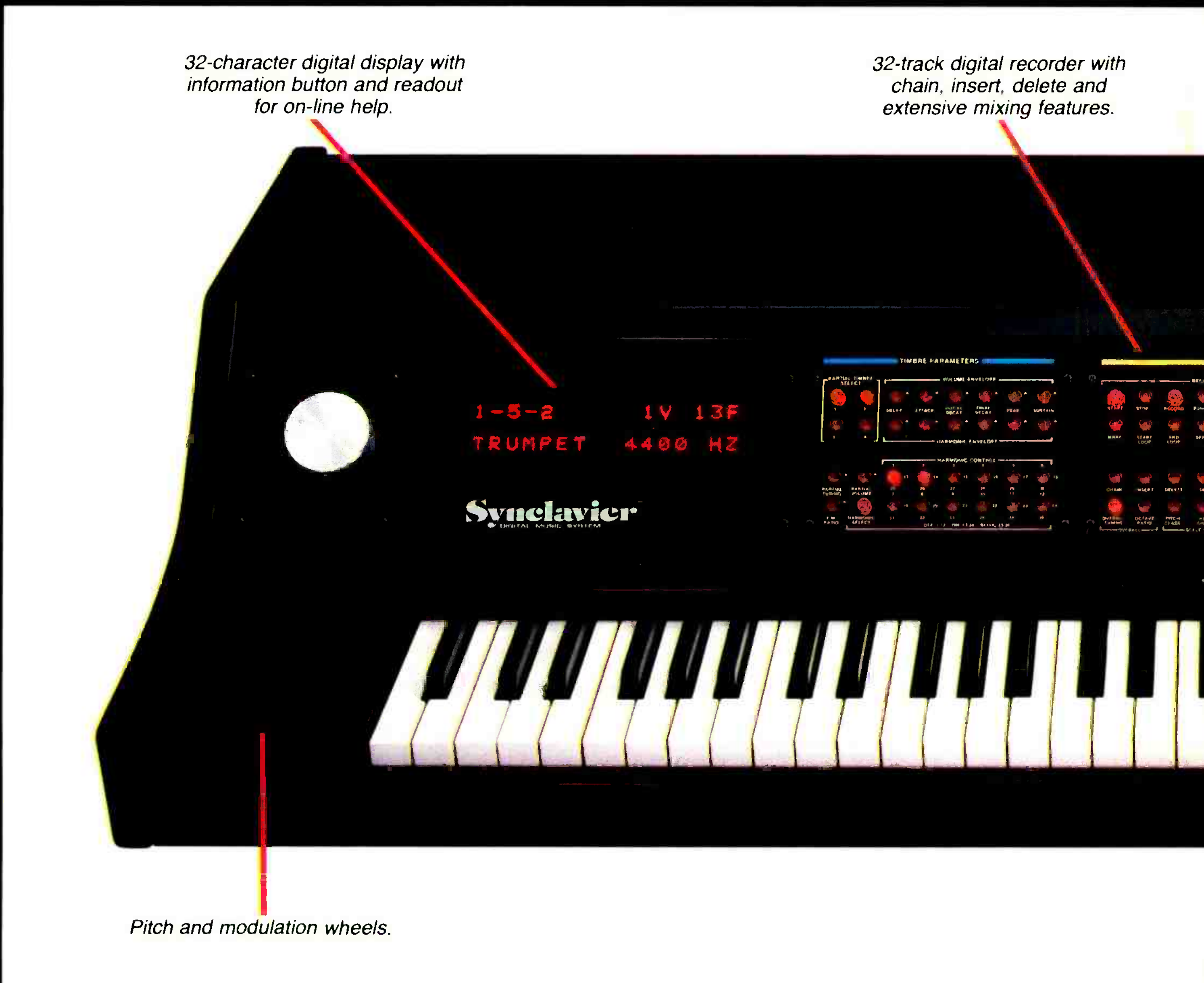

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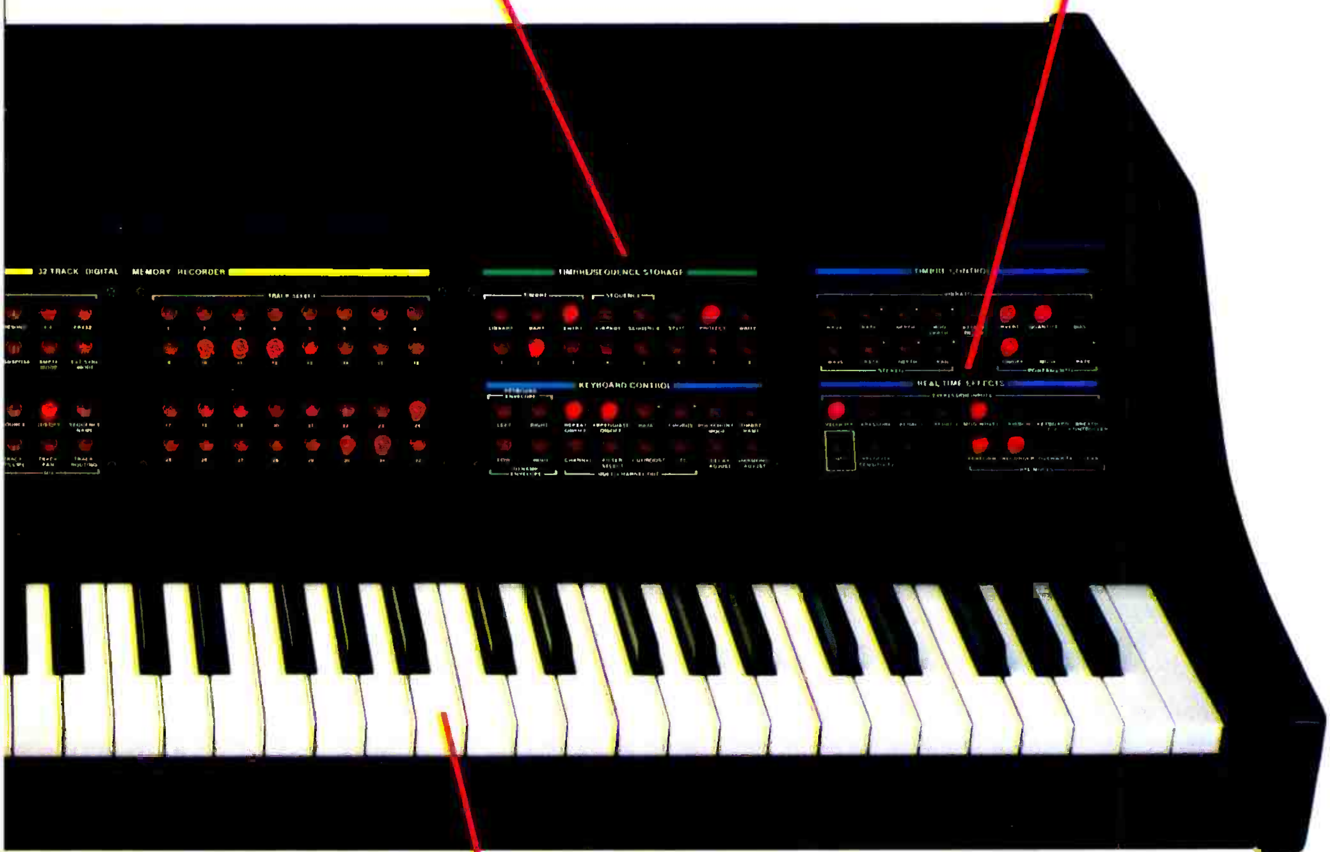

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mixing to your four discrete [unencoded] tracks; you monitor through the matrix to hear what's going to happen later on, when the signal is encoded. The Dolby representative will be there as well to show you the appropriate meter levels, and keep you out of trouble in general."

Equalization Compromises

Along with the problems of creating four relatively discrete channels for the Dolby Stereo format, Prince's two-track mix also presented some equalization difficulties. The main problem that Brody faced was that the supplied mix did not allow for the equalization of any individual instruments or voices.

"From the film mixer's point of view, it would have been better to have the vocals and lead instruments on a separate track. This way, if either a vocal or a solo required special treatment, we would have been able to do it. You generally don't have vocals and solos happening simultaneously, so it would have worked out fine to have them both on the same track. The problem I had with the two-track mix was that I was either equalizing for the voice, or equalizing for the band. I couldn't find one EQ that worked satisfactorily for both.

"Plus, with the vocal or solo instrument on a separate track, we could have been able to duck them under the dialog without having to pull down the whole

track. The vocal or solo would be put on the center track and pulled down to accommodate the dialog — also on the center track. Meanwhile the wings [left and right] could have continued pretty much at full volume."

Equalization for *Purple Rain* was handled on virtually a frame-by-frame basis, according to Brody. He prepared a predub of the music for the film, taking it through a three-stage EQ system. As the first stage, the left and right channels were treated with an Orban stereo parametric equalizer. Here, as elsewhere in the chain, Brody's approach to equalization was to begin his EQ treatment on each channel separately.

"I would generally start with the left channel and get an EQ that I was happy with," he explains. "I would then turn off the left channel, bring up the right channel, and do the same thing. When I brought up both channels together, I generally noticed that I had equalized each one very similarly. This was a good thing, because if I had treated my left and right channels too differently, the center would have come out sounding *very* weird. On listening to the two channels together, I would then make some further adjustments to the EQ.

"I think we got more definition on individual instruments using this approach than we would have if I had just applied the parametric to the stereo left and right channels simultaneously. I wasn't always sure of where some things were coming from just listening to the left and right together."

Brody then brought the left and right channels through the four-band equalizer fitted to Glen Glenn's custom ADM console. "I added a little subtler EQ with the board equalizers, which were fairly complete. A four-band EQ is useful for treating an entire group with vocals, because you can use individual bands to tune in on certain parts. I found I needed to add a lot of bottom to the mix; it seemed to need a lot of warmth. The board gave the mix the extra fullness it needed."

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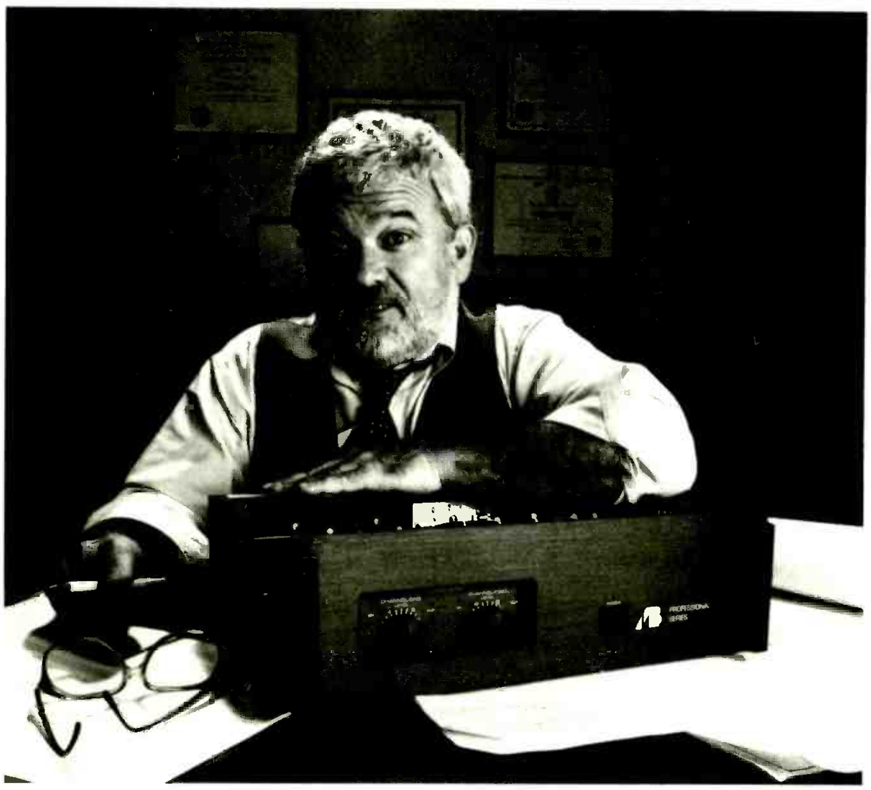

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Immediately after this EQ stage, Brody put the signals through an Aphex Aural Exciter to achieve more of a "lively" sound. The director, he explains, "was emphatic about keeping alive the energy of Prince's original mix. The Aphex was helpful at this particular point in achieving that goal."

The final stage in the predub equalization was to take the left and right signals through two six-band graphic equalizers, also on-board at the Glen Glenn ADM console.

"With the graphic, I was able to vary the EQ smoothly," says the engineer. "With a parametric equalizer, you've got too many variables to effectively change settings during a song. For each of your bands, you've got EQ range and level to consider; it's *too* much to be fidgiting with during a song. It's a different situation with the graphic. If the voice was too harsh in a certain section of a song, for example, I could pull down the appropriate frequencies to soften it. The nice thing about a graphic EQ is that it's not noticeable when you're making changes."

Further equalization was added after the predub, during the final mix, as Brody explains: "When we did the final mix, I still went through the four-band and graphic equalizers on the board one more time. On a couple of the more troublesome selections, I even went through the Orban again for spot EQ. There was one song called 'The Beautiful Ones' that gets to a point where Prince starts really screaming. It's very dramatic, but also very hard on the ear. I didn't want to stifle the dynamic urgency of the song; I wanted to make it sound bright without being piercing.

"The screaming was very peaky in the 3 to 5 kHz range, so at that point I had a setting on the Orban that was a very narrow-band dip at 3 kHz. It was as narrow as I could get it — almost like a notch filter. When that section of the song came up, I just clicked in the Orban. Without taking away from the urgency of the music, the Orban brought the tone down to a point where it was smoother in the context of the rest of the music. We did quite a bit of that kind of spot equalization during the final mix. There were sections throughout that needed special attention."

The area of equalization is one where Brody feels his experience as a record mixer really paid off. For a variety of artistic and economic reasons, record work, he finds, offers more opportunities to study the effects of EQ and other processing on individual tracks.

"A record mixer has probably had more time to experiment with equalization, outboard equipment and certain other things," he states. "Film mixers don't always have the time to surgically investigate individual tracks, as you do in a record mix. As a film mixer, you're often given a three-track composite mix, and the cue sheet may not even note what instruments are on what tracks. The mixer might not have the opportunity to listen to each track before starting.

The other people involved may approach it from an attitude of 'We have a limited amount of time to do this, and time is very expensive; so just make it work.' Sometimes the only way to get an idea of what is on the individual tracks is to push up each fader during a rewind at normal speed and listen to each track through the monitors.

"A record mixer's experience is just the opposite. In a record mix, you can spend hours just playing with a bass-drum track. Based on the amount of time he has spent doing that, a record mixer has an advantage in being able to arrive at an EQ setting rather quickly in a film context. He might be more familiar with what a bass drum sounds like, than a guy who has only heard a bass drum track as part of an overall percussion track.

"On a lot of projects like *Purple Rain*, where the people involved are record people, they insist on the same kind of mixing considerations that they would have on their records. It's not unusual for them to come in with a 24-track tape — something most film guys have not had a chance to really dissect.

"Working from three- and four-track music masters, films mixers are more akin to record mastering engineers in what they do. They have an existing

. . . continued on page 36 —

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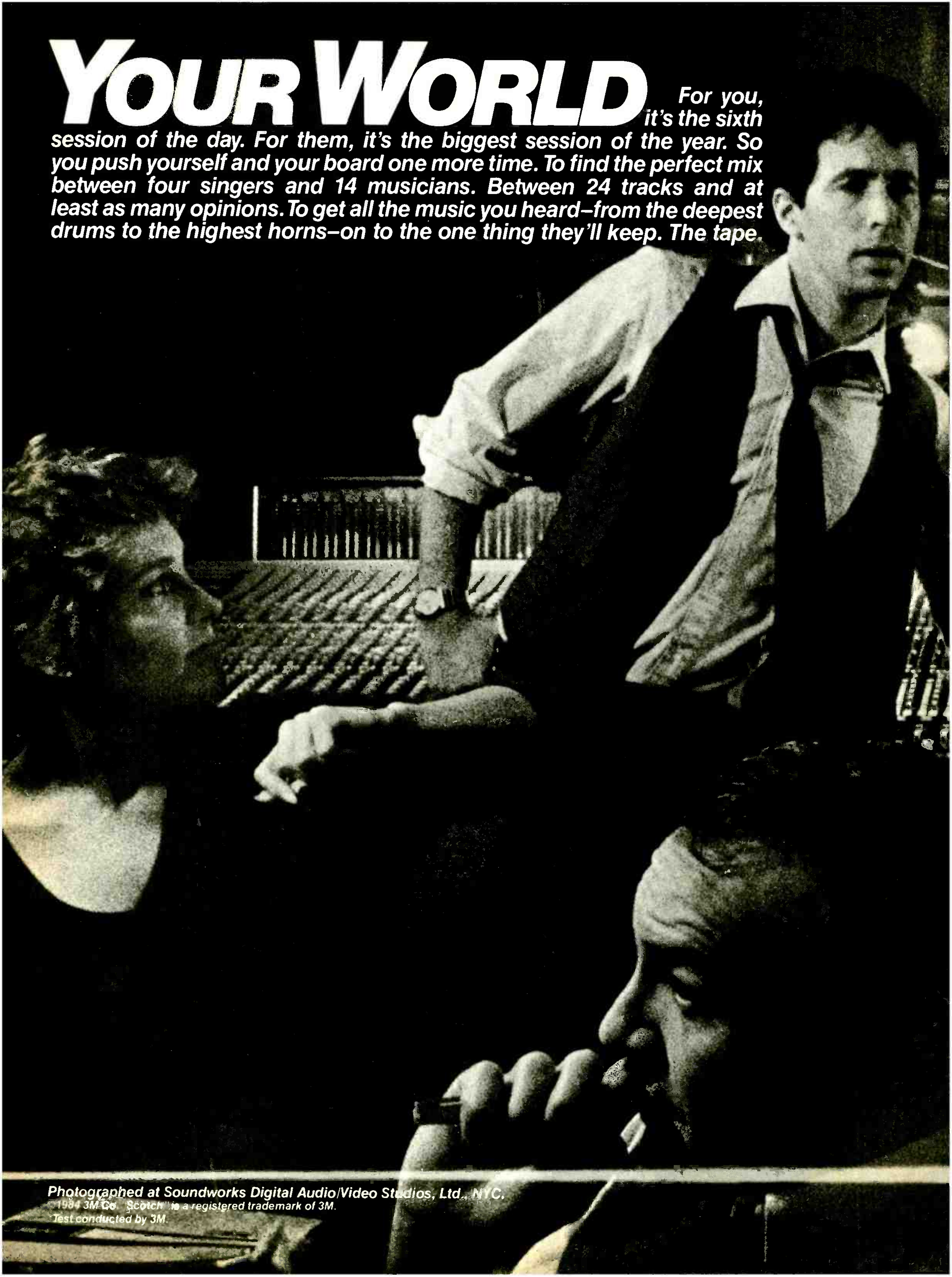
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— *continued from page 32* . . .
mix that they put the finishing touches on."

In context, the film mix of *Purple Rain* may perhaps be seen in terms of disc mastering — a field, interestingly enough, in which Brody began his career. Few disc-mastering projects, however, involve the intricate and varied equalization called for in this instance.

"It was a very complicated project," Brody concedes. "With just a two-track to work from, it sounds as though it would have been fairly easy. But it was complicated in the sense that we had to take every section of music almost as an individual piece. Our EQ was virtually based on whatever we were hearing at that particular moment. I understand, though, that Prince was pleased with the final product on film, so I guess we did our job."

Although it presented special problems of its own, *Purple Rain* is indicative of a new musicality in feature films that extends beyond "pop music movies." Brody sees a growing emphasis on music in all of the film projects that come his way. "There seems to be more of an emphasis in *all* films these days on making the music more integral to what is happening on the screen. Take, for example, a film called *Grandview, USA*, which we recently did here at Ryder Sound. Though not a 'rock movie,' it had many MTV-like scenes where music played a very central role. Whether he comes from a film background or a record background — and there are some excellent music mixers from both areas — the music mixer today tends to have more space to work in than he might have had before."

■■■

News

AMEK REPORTS SALES SUCCESS OF M3500, M2500 AND ANGELA CONSOLES

Notable contracts reported by the company, either completed or in work, include a new M3500 console for **Craig Huxley**, Los Angeles, configured 56/48 with complete 48-track routing and monitoring; a 56/48 M2500 console to **Paramount Pictures**, Hollywood; a 48/48 M2500 for **Craig Huxley**; a 48/48 and a 36/24 M2500 for **Post Sound**, Los Angeles; a 36/24 M2500 for **Music Annex**, San Francisco; a 36/24 M2500 for **The Ranch**, New York; a 42/24 M2500 for **Livingston Studios**, London; a second 26/24 M2500 for **Musicworks**, London; a 36/24 M2500 for **TV Ashi**, Tokyo; a 48/48 M2500 for **Jive Studios**, Tokyo; plus a quantity of M2500s for the **People's Republic of China.**

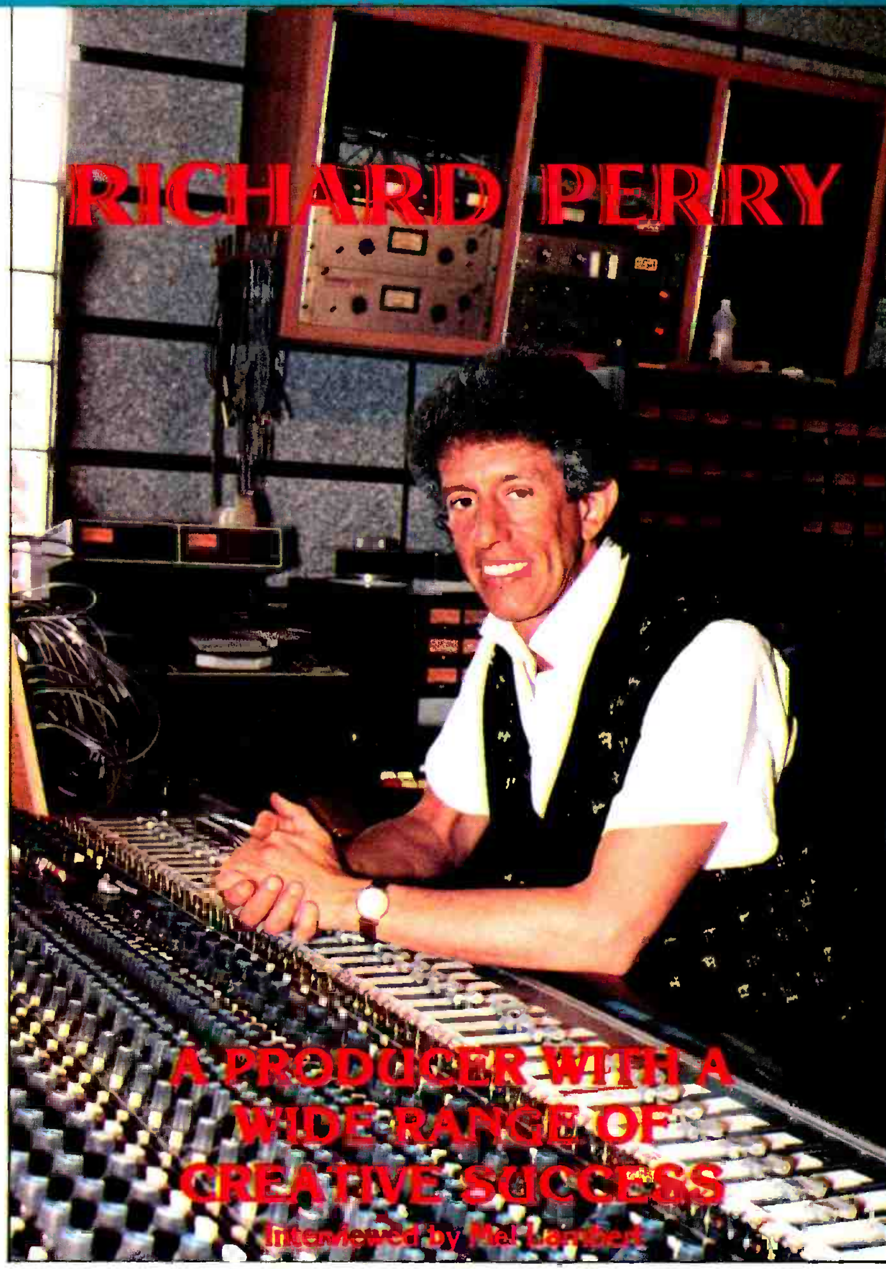
Recent sales of Angela consoles include units to **John Farrar**, Los Angeles, for Olivia Newton-John's new album; **UB40**, Birmingham, England; **Media Recorders**, Hollywood; **Back to the Bible Broadcasting**, Nebraska; **Starsound**, Reno, Nevada; **Mitch Easter**, Winston-Salem; **Mastersound**, Virginia Beach; **Dieter Rieth**, Stuttgart, West Germany; **Jonah Lewie**, London; **New Found Sound**, New Jersey; and **Carnegie Hill Studios**, New York.

M1000 consoles have been sold to **CCTV** (China Central Television); **Paramount Pictures**, Hollywood, for the *Entertainment Tonight* program; **TV3 Cataluna**, Barcelona, Spain; **Mama Records**, Milan; **BEB Records**, Milan; **Channel 5 TV**, Milan; and **Sunny Super Sounds**, Bombay.

STOLEN MICROPHONES

The following microphones were stolen recently from Studio 55, Los Angeles: Neumann U47 FET (#G2780), U47 Tube (#3572), U67 (#33725), U87 (#23085), U87 (#28976), U87 (#28055), AKG C414 (#006), C414 (#1243), C414 (#1201), C414 (#1625) and C414EB (#1179).

All of the microphones were engraved with the legend "Studio 55." David Dubow, 55's studio manager can be reached at (213) 467-5505 for further details. . . . **MORE NEWS** *on page 206* —



RICHARD PERRY

A PRODUCER WITH A WIDE RANGE OF CREATIVE SUCCESS

Interviewed by Mel Lambert

Photography by Kathy Cotter

Without doubt, Richard Perry has amassed an enviable array of production talents during his nearly 20 years in the business. Since relocating to the West Coast in 1967 to become staff producer at Warner Brothers Records, he has been involved with a wide range of artists, including Barbra Streisand, Carly Simon, Harry Nilsson, Leo Sayer, Martha Reeves, Diana Ross, Manhattan Transfer, and many, many more. Having set up his own in-house label, Planet Records, in 1978 — some three years after opening Studio 55 for his personal projects — Perry has seen greatest success in recent years with The Pointer Sisters; at the time of this interview, the *Break Out* album had just gone Platinum. He still finds time for session with non-Planet acts, however, ranging from Neil Diamond and Julio Iglesias, to the title track from Barbra Streisand's forthcoming ablum, *Emotion*. By all accounts a vibrant and creative individual — and possibly unconventional in his future plans to branch into both video and film directing — it is hard to imagine Richard Perry forsaking his first love: Producing Music.

R-e/p (Mel Lambert): For the last couple of years you've been somewhat out of the production spotlight. The impression I get is that prior to the Pointer Sisters' album Break Out, *which has been a huge success for you, you've been adopting something of a low profile. Was that intentional? Have you been preparing to enter another stage in your career?*

Richard Perry: Not intentionally, or consciously. When I made the decision to form Planet Records in 1978, I knew that I was going to be making considerable sacrifices by taking myself out of the mainstream of independent production, and concentrating on the Pointer Sisters and other artists on my own label, plus several new artists. During that period of time, we were really trying just as hard to make the right records then, as we are now. But I've always been a firm believer in fate and everything happening when it's supposed to happen, in its own time and place; it just seemed that this was apparently *the* time.

I've always been a tremendous believer in continual growth. Even though I feel that I know a lot from having been doing what I'm doing for 19 years, I always take the attitude that I still have as much to learn as anybody. I don't think that any of us can feel that we're too big to grow — that is really one of the keys to why, in 1984, I'm having this kind of success with the Pointers.

Music is really changing in a way that has been extremely exciting to me. A couple of years prior to my recent successes, when I was going through, as you described, that low-profile period, music was still redefining itself; the technology was changing very rapidly — new synthesizers, drum machines — and different techniques were coming into play. Now we are starting to rediscover the value of the song.

When I started to do this album, *Break Out*, I put together all the information I'd taken in over the past few years, and was very fortunate for it to happen. It could just as easily have not happened; that's the mercurial nature of our business.

R-e/p (Mel Lambert): I noticed from the album credits that several songs were associate-produced by such people as Glen Ballard and Brock Walsh ("Automatic"), Gary Skardina and Stephen Mitchell ("Jump (For My Love)") and Andy Goldmark ("Telegraph Your Love"). How does the role of associate producer translate to those sessions?

Richard Perry: That's a good question. I credited them on the album simply because of the fact that one of the effects of the changing technology is that now virtually any writer, working in the comfort and privacy of his own

"Rich is good at hearing where parts should go without stepping on other things. Everybody talks about melodic and harmonic rubs, but there are 'sonic' rubs — piling too much in a given frequency of the total range that you hear."
— **Stephen Mitchell**, session musician and co-writer of "Jump (For My Love)" from the Pointer Sisters' *Break Out* album.

RICHARD PERRY

home with just a drum machine or synthesizer, can make a demo that has as good a sound as most people would have gotten up until just a few years ago in the studio. As a result, a writer can create a demo that is considerably closer to the master than in previous times.

I've always been a strong believer in capturing the magic of the demo, because there's a very *special* atmosphere that takes place when you do a demo; it's completely pressure-free, and totally creative. But, when you go into the studio and do the record, even if you have the best musicians in the world, a certain amount of tension creeps into it. Many times it would be a real struggle to recapture that magic of the demo. With all of these new drum machines and synthesizers, you can listen to the demo track and say, "Well, this has a great feeling; it's got all the basic elements that I would want to incorporate." And then build from there.

R-e/p (Mel Lambert): So you would encourage a writer to take on that role of associate producer, because of the sounds they might develop for a demo?
Richard Perry: It wasn't even a question of us talking about any specific roles. The writers were delighted to participate in that capacity and, working with me, to be able to ensure that their songs were going to be made in the same creative environment and spirit that they created them. Sometimes we would take the same track, and then build and expand on it from there; otherwise I'd be virtually duplicating their efforts. Now I can just get the writer to program the drum machine, for example, in the same way he or she had set it up on the demo.

Here again, we have to stop and really think about the technology element. If it's something as simple as just putting the same program in the same LinnDrum machine, what's the point in changing it? For instance, if you could listen to the demo of "Automatic," you would hear the *exact* same rhythm program, and so on and so forth. Obviously, you'd hear some substantial differences; the record has grown and been fleshed out. Sometimes major differences can take place, even if you start with the same basic track.

R-e/p: Then, as the producer, you will take that basic sound as an element, and suggest that other textures and sounds could be added to it?
RP: Exactly. That's the beginning element; the seed. Everybody knows there's still a lot more that goes into the making of a record, but it's a vitally important first stage. Because the writers I worked closely with on *Break Out* all gave of their time so generously and openly, when it came time to putting it all together I chose to credit them all as associate producer on the individual tracks. Another person that I had discovered in the course of working on this album, Howie Rice, was credited as an overall associate producer. Howie came in on every track, and added a lot of very important musical elements — no matter whose track it started out to be — as well as working on some tracks he built entirely by himself.

R-e/p: You were taking input from a lot of different people on those sessions?
RP: Yes. I've always been open to the best idea, wherever it comes from. Then the most important thing is being able

to filter the various ideas, and be able to inspire them toward the best possible solution.

R-e/p: Currently you are working with Barbra Streisand on her new Emotion *album. I know you worked with her on the* Stoney End *album way back in 1970. Is this the first time you've worked with her since those days?*
RP: Actually we did three albums back in the early Seventies; *Stoney End* was the first, recorded in 1970. Then we did an album in '71 called *Barbra Joan Streisand*, and, in 1972, a live album, *Live Concert At The Forum*. Since that time a lot has happened in both our lives, and we've remained friends all that time.

She had just come back from doing *Yentl*, which represented an obvious milestone in her career, and was going through a period in her life where she wanted to just relax now that she had that behind her. So we started to spend more time together as friends; working together with her wasn't anything that I particularly thought about.

She just mentioned to me a couple of months ago that she was getting ready to do an album, and was having different people produce different tracks. It really meant a lot to her to have me involved in it, so I started looking for a song and came up with "Emotion," from a writer in New York, Peter Bliss. She liked the song, we recorded it, and then subsequently made it the title of the album. [*Emotion* is scheduled for release in September — *ML*.]

Nothing's better than working with a friend, and the working experience was even more delightful than ever. We're both very pleased with the results; it's a very exciting record, and very different for her. Recently I added the Pointer Sisters singing backgrounds with her, which is a very exciting sound.

R-e/p: Last year you recorded some tracks with Julio Iglesias. Subsequently we heard that the album, 1100 Bel Air Place, *had passed to another producer. How did that come about?*
RP: Quite simply, Julio felt that he needed an American producer to pave the way for him if he was to break into the U.S. market; he couldn't just do it with his Spanish producer. CBS had been wanting me to get together with him for five years, but until now it didn't mean enough to him to have to sacrifice control. Finally he decided he wanted to do it, and it was all put together very quickly. We hit it off really well personally, and I felt that that would be enough momentum to carry us into the working relationship.

I spent a year working with him, on and off. During that time I recorded about 16 tracks, of which I discarded several that I felt were too blatant attempts to make him sound commercial. We were experimenting with some "Flashdance"-type songs. I realized that it was stretching him too far. You needed to keep him true to his style; that sensuous Julio Iglesias mood . . .

R-e/p: . . . The Latin Lover?
RP: Right. So I reduced it to what I felt were six very strong tracks, of which we needed four more. Of those six, one was the duet with Willie Nelson ["To All the Girls I've Loved Before"], which I recorded; the Diana Ross duet ["All of You"], which I don't think would have ever happened if I didn't have a longstanding relationship with Diana; and the Beach Boys' track, "The Air That I Breath," which is a *very* strong track. I also brought in Stan Getz, who Julio really wasn't that familiar with, to play solo on one track that he subsequently discarded, and then brought Stan back again to do some new tracks.

But more important than that is the fact that I had really shown him the way of recording in Los Angeles and working with new musicians. I worked hard to bring Albert Hammond into the project; he had an old standing relationship with Julio, and spoke fluent Spanish as well as English. I also introduced him to Humberto Gatica. Surrounding him by Spanish-speaking people was very important, I felt, because subsequently the language barrier was a very substantial obstacle.

R-e/p: Julio doesn't speak English at all?

RP: He can speak English and is a very enchanting, charming man, but when you get involved in the work process in the studio, communications become exacting. And even an interpreter doesn't do the trick, because no one could possibly interpret my emotions specifically. I consider my ability to communicate the ultimate strength of my talent as a producer, and having that ability severely impaired made it difficult.

Apart from the fact that Julio, unbeknownst to me at the time we started to work together, has never relinquished control of his records, and spends an inordinate amount of time making them, even in Spanish.

It's long been water under the bridge to me, but I think quite honestly that Julio felt that he could take it over on his own, now that I had surrounded him with all the necessary tools he needed to continue the album on his own.

So often with artists it isn't until the album is totally put together that they realize the whole is greater than the sum of the parts.

R-e/p: To continue with a look at some of the projects you've been involved with recently, at the moment you are mixing the latest album from Greg Phillinganes, titled Pulse. *Greg has worked with Stevie Wonder, served as a session musician, arranger, producer, and is now finished up his new solo album. How did you become involved in Greg's music?*

RP: I signed Greg as an artist to Planet Records a few years ago, and during that time we did a lot of talking about potential directions for his recording career. He was very involved in other people's records, and went on tour with Lionel Richie for four months as his musical director. I was in no rush, and neither was Greg, because we both knew that when the time was right it would all come together.

No matter how talented an artist may be, if they can find the right person to work with — one they can trust with their music — they're always better off *not* having to produce themselves. On this new album we worked very closely together. Greg would do a lot of the synthesizer overbubs and various things in

the studio by himself, but I was always very close at hand to give comments and guidance.

I think the album is going to be very, very strong, and one that we're both going to be very proud of. There's an incredible song on the album by Michael Jackson that he wrote some years ago with the Yellow Magic Orchestra, called "Behind the Mask." There's also a song by Jackie Jackson, "Playing With Fire," which is extremely strong.

R-e/p: You also cut a couple of tracks on the new Neil Diamond album, Primitive.

RP: That was done last January, when Neil was completing his album. I have known Neil for several years, and we'd always wanted to work together. He had written some songs with Burt and Carole Bacharach, who are very dear friends of mine. There was maybe a week left to get some songs on the album, and they wanted me to do two tracks. So I picked two of the songs that they had written together — "Sleep With Me Tonight" and "Crazy" — and

we went ahead and did both of them in a week.

R-e/p: Do you think that having a restricted time frame of one week to record two important tracks helped to clarify what you were doing in the studio?

RP: That's a good point, because I enjoy working under pressure . . . up to a point [laughter]. I *wouldn't* have wanted to do the whole album in four weeks, but it was okay. Just so long as you have the ability to come back and remix — it's important to be able to come back to it again with a fresh ear, if necessary. But we're all very pleased with the results. Neil is one of the most cooperative artists I've ever worked with; he gave me complete, free rein to do my job, and was there to add whatever support he could.

R-e/p: When you are asked to produce just one or two cuts for an album, do you ever listen to the other material that may already have been recorded, or do you need to retain an independent viewpoint?

RP: Until this year, I've never worked on an album where I did less than the whole project. I'm not particularly fond of it, conceptually, because I always like to think that the tracks that I do will be the hits and, as we know, that is really what's going to carry the album. But certain artists do seem to be using several producers on an album, and I said to myself that, even though the remuneration may not be completely just, there are things more important than money. Not that money has ever been a motivating factor for me, but there are one or two times when you just say to yourself — just from the standpoint that it's going to be fun — what the hell let's do it!

And it gives you the chance to work with somebody without having to be involved in the *whole* album. For instance, I wouldn't have had time to do Barbra's entire album, much as I would love to do that more than anything. But I *did* have time to do the one track. It was the same thing with Neil Diamond.

I wouldn't want to do this as a normal practice, however, mainly because the one thing I enjoy more than anything is the ability to creatively craft the album — that's what get's me off more than anything, as opposed to having my one track mixed in with the others.

But to get back to your question, in both those cases, I didn't hear anything that had been done prior to my tracks, nor did I have any particular desire to. I felt very secure and focused in terms of the way I go about finding the right song and making my record, and that's what I was hired to do.

R-e/p: We seem to be getting to the heart of the producer's role. In essence, how would you sum up the role of a producer? Does it start with choosing the right songs for the artist; songs you can believe in?

"Most producers [that I've worked with] keep their fingers on every aspect, which kind of stifles you. Rich will tell me where he needs a solo, and gives me a general direction. He feels comfortable in just saying: 'Let me know when you have something you like'."

— **Michael Boddicker**, synthesizer player on "Crazy" and "Sleep with Me Tonight," from Neil Diamond's album, *Emotion*.

RICHARD PERRY

RP: Yes. From my very beginnings in making records — and I say it today more than ever — the song's the thing. A lot of producers don't have a particularly good song sense, but all the technology in the world means *nothing* compared to the song. I've heard a lot of hit records that sound horrible, in terms of the actual production, yet they still can be hits. Occasionally, a dance-oriented type record , which is not what you'd call a great song, can still achieve a measure of success on another level.

Not only does the producer's role come into play finding songs, but I work very closely with the writers, sometimes re-writing melodic or lyrical sections. Being able to help them to make the song as great as it can possibly be, is one of the most important elements that I'm involved with in the making of a record.

R-e/p: Another facet of the producer's role is to allow the artist to give the performance that you know they are capable of giving. Does that come down to setting the right environment, selecting the right studio, the best musicians?

RP: There's no question about that; to really understand the psyche of the artist, in terms of any creative collaboration, is *vitally* important. So many people tend to overlook or underplay that aspect, because it's not just a question about coming in and saying, "Okay, let's do it." You have to be able to truly inspire an artist by setting the most conducive mood possible.

When I've recorded vocal performances with an artist, for instance, let's say we go back in the studio two or three times to take different shots at the lead. If I'm doing my job right, each take should be like a major step forward, because of the continual turn-on of the way they relate to a song. Mind you, I also love to get live vocals; I'm not a big believer in doing *endless* vocals. But again, it all depends on the artist, the song, and the circumstances. You have to have the sensitivity to know that you've gotten your best performance, or that your best performance is still ahead of you, and if so, how do you inspire the artist and get to that point.

R-e/p: I notice from your biography that you have a degree in music and drama — an odd combination. Has that mixed background helped you in your career?

RP: I think that it has, although of course none of it is an *essential* requirement for being a record producer. Having that kind of musical training, in terms of learning how to play practically every instrument, has helped in the communication of music, which obviously is so important. Especially in the old days, when we used real instruments. When I would do a string session, maybe 16 or 17 years ago, if a particular passage wasn't being phrased properly, I could give bowing instructions to the musicians. Or being able to tell horn or woodwind players how to tongue a particular passage, and get the desired effect I wanted.

In terms of my drama background, an intelligent producer has to play a lot of different roles. To me, what makes a brilliant actor is the choices he makes to get himself into the particular role.

R-e/p: It has been suggested that more and more the role of the producer is akin to that of a chameleon — the ability to merge with the session, but still retain that overall responsibility, which everybody expects the producer to provide.

RP: The role of the producer is also analogous in many ways to that of a film director. But that analogy is only applicable to certain producers. After all, there are some film directors that I happen to know for a fact will say something on the set like, "Be funny," which is obviously the last thing in the world an actor would be able to do on command.

I think the chameleon example is a very appropriate one but, beyond that, it's being able to recognize the most effective role for you as the producer to play at that moment.

R-e/p: So being a producer comes down to understanding people; their motivation, and what makes them tick?

RP: Precisely. Being friend, parent, teacher, therapist, peer . . . so many things. The first thing I do when I start to work with an artist is to assess their strengths and weaknesses. No matter who the artist is, they look to the producer to play a variety of roles. You have to size up the artist's psychological make-up, and know what turns them on and what turns them off. And, when those various elements come into play, be able to push those buttons, make those changes precisely at the time that they're necessary.

R-e/p: On the technical side, how do you set yourself up for the recording process? Do you go through a rehearsal and pre-production stage? Do you look to set a target date for recording basics in a certain period of time, and leave a reasonable amount of time for overdubs, vocals, and mixing?

RP: It's pretty much the way you described it, although not quite as rigid. With the exception of something like the Neil Diamond situation, where I discovered I had only a week to do the two tracks, there's no specific time schedule. I know, for instance, that if the Pointers are only going to be in town for two weeks I'm going to be doing basic tracks for that period. I'll organize it so that maybe I'll do basic tracks Monday, Tuesday and Wednesday, some rough vocals on Thursday and Friday — depending on how good the live vocal is

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THE ROLE OF THE FIRST-CALL ENGINEER
A Conversation with Richard Perry's current session engineer, John Arrias

Independent engineer John Arrias recently joined Richard Perry as his first-call engineer, and will be joining the producer on his upcoming projects. As Perry explains in this month's interview, he is looking forward to working again with one engineer from the tracking dates, through overdubs, to the intricate remix stage. And Arrias indeed seems capable of fulfilling Perry's exacting requirements for an all-round engineer. During the past 10 years he has worked on numerous landmark sessions, including Barbra Streisand's *Wet* and *Memories* albums; *Stranger in Town* with Bob Seger; Donna Summer's "Enough is Enough" single; Jeff Beck's *Wired*; Hall & Oates' *Bigger than Both of Us*; *Thelma* with Thelma Houston; *Feets Don't Fail Me Now* with Little Feet; plus albums and singles with the Marshall Tucker Band, Rufus, Lynn Anderson, Stephanie Mills and Chaka Kahn.

How does Arrias lock himself into the mood of a Richard Perry album project, we queried?

"So far, I've done very little recording with Richard; it's mostly been mixing," Arrias concedes. "On the recent Greg Phillinganes project, he gave me rough mixes that he's particularly liked the feel of. There's a song on *Pulse* that was written by Donald Fagen, so I listened to some Steely Dan tracks to try to cop that kind of attitude.

"Prior to mixing, Richard will talk to me about a particular project. He usually puts on tape the sound he really likes to hear; he doesn't like to alter too much of the sounds he captures in the studio — it's more a balance and 'color' changes in a song. In listening to the rough mixes, Richard tries not to give me too much of a guideline, because he wants to get my fresh input on a mix.

"So I listen to the song, start to bring up tracks, and hear things that maybe he let pass. For instance, there are a couple of tunes on Greg's album that he let me mix by myself to see how I heard them. And he called me and said: 'Gosh, there was a guitar on there that I didn't he even know existed.'

"Sometimes he comes in and says, 'Wrong, you've taken it *too* far.' For one particular song on Greg's album, I took it into more of a 'techno-type' feel. There were a lot of synthesizers on the track, and a lot of aggressive drum sounds, so I really attacked it and came up with a lot of extremes. Richard came in and said, 'Too much. Tone everything down; make it *smaller*.' I'll then bring it way down, and start bringing back the gated echoes and things I was using, plus trying different chambers, to tone down the whole song. Then he says, 'Now rebuild it; this is what I want. Reconstruct and build it up a little bit from there.'

"Then Richard gets a little more technical: 'Drums are a little too loud; I don't like that guitar sound; I want more of the percussive feel; the horns need to come out a little more —I need those to *pop* at you.' After that he'll join me behind the console, and we'll work on each individual track. Because he has an overall concept of the song, Richard likes to try some balances. Then he'll leave the room and I'll smooth it out a little bit — he might be a little erratic with the balance because he's trying to recreate a particular feel on the tracks.

"We'll continually refine the mix, and then throw questions back at each other about different aspects. 'How about these horns coming out? I'd like to hear the swell of them a little more.' We'll bounce ideas back and forth.

"Maybe while mixing we might discover an element that doesn't work, so we'll bring in a musician to overdub a part straight into the mix. Sometimes you have been listening to a

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— and then add certain textures after the basic track.

A good example of that was when we went down to cut the Willie Nelson and Julio Iglesias record. I recorded the basics the night before, including putting on some quick synthesizers, because I wanted the artists to have something more to sing against. Sometimes those kind of sustaining colors can be very important, even if you wind up changing it afterwards. What you give the artist to sing against is going to have a big effect on what inspires them; it most definitely triggers in the emotions.

Having done it for so long now, and having my own studio, I don't have to clock in every day, but I have pretty much of a feel for how it's going, and where I want to be at certain times.

One of the interesting elements of the latest technology is that it does give you a marvelous option of making changes at the last minute. For instance, on "Automatic" I changed back-beat sound just before I mixed it, because you can just hook the Linn drum machine to a sync pulse, and add whatever you want. Which is like performing open heart surgery; you would *never* dream of changing live drums once they're locked in the groove. But now you can quickly make a track that sounds good, get your vocal on it, start to build your record, and then replace some parts, if necessary. To be able to do that, and know that you are not in any way tampering with the feel, is a marvelous option.

R-e/p: What do you track as basics during an "average" session? Do you try and put as much as you can down as possible?

RP: Not necessarily; it all depends on the date. If I'm talking about working with live musicians, I usually go for drums, bass, one or two keyboards, and usually two guitars. I like to get the thrust of the rhythm session as complete as possible. I've also done a lot of tracking with percussion; sometimes it can be really good for the feel if the percussion is going to provide a particularly important element that you want to be able to interact with the other musicians on the basic track. But I have found that nine times out of 10, it works better by overdubbing the percussion; even when I've done it live, a lot of times, I'll come back and redo it as an overdub.

R-e/p: How have your techniques changed now that you're working more and more with electronic instruments? Do you lay down a click-track first to provide a rough sequence of drums?

RP: Usually we'll put down the bulk of the drum part, and a synthesizer bass. My intention is to get the track more or less together fairly quickly, and then to put a vocal right on it. This gives you a great opportunity to hear the sound

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without having to go in and cut a whole track and then discover that it's not happening. In a matter of a few hours you can have a basic track, put on a vocal,and be sure of whether you're getting the magic from the combination of that artist and that song.

R-e/p: At that stage you can hear whether or not the tracks are going to work?

RP: Yes, I can usually hear it right away. That's not to say that there have been many records I've done where maybe a key overdub suddenly forms the catalytic element that pulls it all together. When I was doing "You're So Vain," with Carly Simon, we knew it was a hot song. I cut that track three times, but it wasn't until I overdubbed the opening acoustic guitar that the record became whole. It's amazing how one overdub can suddenly bring it all into focus.

R-e/p: On the engineering side, do you tend to leave mike selection, placement, and that sort of thing to your engineer?

RP: Yes, I like to leave as much of the technical side to the engineers. Back in the late Sixties, like most producers, I pretty much left it all to the engineers. I didn't know any of the *technological* terms, but I knew *music*; I knew what I wanted. I had to describe it to the engineer in my own words, and then be completely at his mercy for which knobs he was tweaking. And if you have a great engineer that certainly is an effective way of working. However, I started to become frustrated at having to be completely at an engineer's mercy, so I started to learn about the technology.

I would observe how they would EQ different things; the effect of different frequencies on different instruments. I started to really become aware of the whole spectrum of frequencies, and was able to hear in my head what a certain frequency would sound like on a certain instrument.

I think that it's very important to have a background in the technology, but now, having almost gone the full gamut with it, I prefer to remove myself from it and only get into it if absolutely necessary. I always look to find a musical engineer, so I can bounce an opinion off someone who is aware of performance, and has a feeling for the song. By the same token, it's great for a producer to have a feeling for EQ, and to suggest to the engineer that they hit this instrument at 3.2 kHz instead of 5 kHz, or whatever.

R-e/p: Rather than saying: "I'd like it a little more edgy, or make that reverb sound blue."

RP: Precisely. Just as the engineer might say, "That was a good take on the guitar solo" — I welcome that kind of input. While we each have our specific functions, I like to make it as much of a collaborative art form as possible.

R-e/p: I've observed that an increasing number of engineers are specializing in different facets of the recording process: tracking, overdubs, and mixing. Sometimes they may work on just a couple of songs for an album; on several of your projects you've used different engineers for different tunes. Bill Schnee, in particular, is credited on several albums as the remix engineer. Are there particular skills that certain engineers possess, and which would make them better suited to one project and not another?

RP: Ideally, there is no greater luxury than having an engineer to work with you from the beginning through the end of a project. But there are certain engineers who excel at recording, and there's a whole other kind of talent and ability that's required of a mix engineer. Some engineers can cut excellent tracks, but may not be great mixers, and *vice-versa*.

The early and mid-Seventies, when I worked primarily with Bill Schnee, was pretty much the last time I worked extensively with an engineer from the recording straight through mixing. I also did some with Howard Steele in 1977, but mainly in the last six or so years, I have given two or three young, up-and-coming engineers an opportunity to work with me. Many were trained at my studio, working their way up to seconds, then given a shot to be my recording engineer.

But I didn't feel that any of my seconds were ready, for whatever reason, to mix on the all-star team. Frequently I would bring in Bill Schnee who, at that point, was involved in his own projects and running his own studio and he really didn't have the time to work on the whole project. However, we were cutting good tracks, so if I could bring in someone like Bill, who's one of the master mixers and can bring all the elements together, it worked out well for me. Also, unless you are working with an exceptionally talented engineer, having a pair of fresh ears can frequently be an asset.

In other words, I'm not saying that the same engineer will *always* mix the track he cut, even if he's a top engineer, since getting a fresh perspective on

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mix so often that you start to tune things out. When you get to the mixing stage, however, some of these elements start appearing, and you discover that the parts do not work together very well anymore. Consequently, in the recording stage Richard will try to record, for instance, three passes of a conga, and will keep all three takes. They'll be three different performances: one might be a total 'feel' performance by a musician; the second with a little bit of direction; and the third one with a *lot* of direction — the specific parts that Richard really would like to hear. We'll decide which one to use in the mixing stage, because one performance might not work with the different textures that came up later."

Drum and Vocal Recordings

Although his tracking experience with Perry is still somewhat limited, Arrias did work on drum overdubs for the Greg Phillinganes album.

"That's the first time I've had a chance to record drums at Studio 55," he explains. "It was a great opportunity, because there were no other musicians on the date and I had total control on the drum sound. On that song, with a drummer by the name of Carlos Vega, Greg needed a large drum sound — and Carlos plays aggressively —so I used a lot of ambient mikes. I gated the snare and bass drum with different noise gates,and used an AMS [RMX-16 digital] reverb for some of the room ambience.

"It's a funny thing how sometimes you take a live drum sound, and try to create a 'synthetic' drum sound, because your ears get accustomed to hearing drums in a certain way. I've recently been working with another producer who uses a lot of drum machines, and I've been getting used to hearing a clean and precise drum sound. When you then hear leakage in a room from a live snare and bass drum, you want to get rid of it. You gate the drum tracks to remove the leakage, but *then* you start losing the ambience. So you bring in the ambient mikes . . .

"You can create the large drum sound with reverb; live tom-toms help, and a lot of drummers are using Simmons drums these days. But artificial reverb will *never* sound like a live room. There are very few good live rooms. Cherokee Studio A is excellent, and Studio 55 is another excellent sounding old room with a lot of wood and parquet floors. But I like to start with the ambience in a room. Then, if I need certain effects on drums, I use a gated-type echo, rather than echo as ambience. I'd rather use the space and decay in a room for ambience.

Arrias has also been recording vocals for Barbra Streisand's new album, *Emotion*. Was her voice difficult to record, we asked?

"No, Barbra's a pro," the engineer enthuses. "I use a [Telefunken] M49 on her voice; she rides the microphone perfectly, unless she's not hearing it properly — headphone balance for her is *most* critical, because she has to hear the echo properly.

"I remember recording Barbra for an album [*Wet*] a couple of years ago at Capitol Studio A, live with a 55-piece orchestra. Barbra was having the hardest time trying to sing, but she

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something can be very important — sometimes an engineer can get very burned out if you don't have time to take a break between your recording and mixing.

But optimally, there is nothing better than working straight through with the same engineer; he knows the tracks intimately — the little nuances — and what went into getting the performance. I've always considered it a tremendous benefit, and one that I have not experienced in some time. I'm hoping to explore that kind of relationship with my current engineer, John Arrias.

R-e/p: From talking with him, John appears to have an impressive track record, and to have been involved in the industry for many years. Will he now follow all of the recording stages from tracking to mixing?

RP: That's our plan, and why we formed this relationship. I feel I'm the quarterback, but I have no great receivers to pass to. You *have* to have someone who can really hold up their end on the same level, because it takes a lot of pressure off the producer.

R-e/p: I'd like to focus on one or two of your recent productions. Let's take The Pointer Sisters' "Automatic" track from the Break Out *album. It sounds like a rather complicated mix.*

RP: "Automatic" was one of the most difficult records I've ever had to mix. The record has a lot of unusual textural elements in it, not the least of which was Ruth Pointer's voice, which I wanted to be out-front. There were so many elements that I had to become involved more in an engineering-type capacity, and make a lot of critical judgements on EQ choice, and things like that. For a while it felt like an unending journey!

We'd mixed "Automatic" several times and, after this one marathon mixing session, it wasn't until I woke up the next day that I really knew for sure I had achieved *the* mix. It was tremendously satisfying to have been able to do it on a record like that, and then to have so many people come back to you and say how great it sounded.

R-e/p: When you reach the remix stage, do you approach it in visual terms? Apart from establishing the depth and width of the soundfield, referring to maybe sonic "colors" and "shapes?"

RP: Very much so. It's interesting that you should mention that, because I've always related to my work in visual terms, but have never heard anybody describe it to me quite that way. I've always applied it to records that way, and sometimes with very specific visual images; it's almost like creating a video in my mind while making the record, particularly during the mixing process.

When I started to mix *Break Out*, I said to the engineers that is was *not* just a matter of coming up with, for instance, complex echoes and special effects. The "effects" are all on the tape; the remix was a matter of balancing the music that was already there. On "Automatic" there's a kind of symbiotic relationship between the sound textures — what we're talking about, in fact, is the arrangement, which I think is one of the most unique elements of that record. What made the mix difficult was that the arrangement had to blend together in the most effective way.

R-e/p: The vocal line sounds underplayed; there's a lot of power there by it being laid back. Ruth doesn't sound casual, but there's a lot of withheld energy.

RP: It's intense, but relaxed. The key with a lot of great contemporary singers, is that they realize "less can be more."

R-e/p: Let's talk about Studio 55. Why did you decided to open your own facility?

RP: As far back as 1970 — when I realized how much time I was spending in the studio — I began to think about having my own facility where everything could be catered to the way I wanted to work.

Our motto has always been: "The best coffee in town;" sounds are not so great, but at least we have good coffee! [laughter] In fact, in 1974, I came very close to buying Sunset Sound [Hollywood]. At that time, the owner, Tutti Camarata, was thinking of getting out of the business, and it was just before Sunset went through a major overhaul. The deal was there to be made, but at the time it seemed to be biting off a little bit more than I wanted at that time, because it is a major complex. I didn't want to necessarily go into the studio business; I just wanted to have my own studio.

Then, a year later, I was spending a great deal of time at Cherokee, and was going to make a deal with them to take over one of their rooms; they would pro-

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couldn't figure out what the problem was. What it finally boiled down was that reverb wasn't coming back in her headphones in stereo; it was a *mono* foldback system. As soon as we switched over, it all fell into place.

For Arrias, the first major project with Richard Perry will be the new Pointer Sisters album, which the engineer is scheduled to track and mix later this year.

"I'm really looking forward to it," he confides. "I haven't had a chance to work with the Pointers yet, but from all the stories I've heard it's really going to be fun. I've been listening to the last Pointer's album for vocal balances and different textures that Richard likes to hear, and to see where he's coming from."

Would you say that there is a "Richard Perry Sound?"

"Definitely," Arrias replies. "It's in the balancing of a mix; Richard likes depth. Having worked with Bill Schnee for so many years on so many major hit records, the two of them have developed a sense of echo or reverb depth. Richard likes to hear it that way, and it's going to be a learning stage for me.

"To me, the 'Richard Perry Sound' isn't a matter of bringing everything up front and hitting you in the face — it is the *subtleties*. Being able to hear *all* the instruments, but way back in the mix. You might not always hear it, but you'll *feel* it. The song dictates the placement, echo, clarity, and punch. The bottom-end has to be right; he likes to hear the punch.

"Once we reach a stage where we think it's almost there, we start printing on the two-track, and will save all those tapes. After each pass, we'll have a new two-track and a new NECAM mix saved to floppy disk — so we notate on the mix, on the two-track, and on the NECAM. We document *everything*; the assistant engineer is constantly writing. We'll even take photos when we change EQ. And at the end of a mix we'll both give our comments to the assistant, and he'll write those down to document the session as much as possible.

"There's a point you reach with a producer — and it usually takes a couple of years —where both of you are reaching for the same knob at the same time. You get to a stage where your minds are working together on the same project, and are thinking in the same way. It's taken a *lot* less time for me to reach that stage with Richard than it has with most other producers I've worked with. Working with him is a unique experience." □□□

vide equipment, maintenance which, as we all know, is the *key* to running a successful studio.

Finally, I had a fellow who'd been working for me for some time, Larry Emerine, search around town, and he found this studio. It's a very historic studio — it was built in 1940 by Decca Records as a studio for Bing Crosby, since it was the only way they could get him into the studio. He just rode his bicycle over from the Paramount lot, which is right next door.

We spent a lot of time restoring it, and building new control rooms. The main room was completely a shambles, so we put the small room into shape, which now has a new, enlarged control room in the back. I've done quite a bit of recording in that small studio; including a Manhattan Transfer album, and three-quarters of a Leo Sayer album.

During that time we took all our money to restore the main room, Studio A, which didn't open till about 1977. The first track I cut in the big room was Leo's "When I Need You." I guess it's been kind of a good-luck room, because I went right from "When I Need You" to Carly Simon's "Nobody Does It Better" immediately after that.

The back room became primarily a mixing facility and then, about a year ago, just before the *Break Out* album, we completed the new control room for Studio B.

R-e/p: Is Studio 55 rented out to other producers?

RP: Most definitely. Because I'm there a lot, we don't actively solicit other clients, but certainly we're always open to the public, and have been since the beginning. Toto recorded their first album there; both of Stevie Nicks' solo albums; the last Bob Seger album; the last Go-Go's album . . . a lot of different artists have come through here. Certainly, we're totally open to any outside bookings. I even go out of my way to be flexible, so that I don't just block book the studio.

R-e/p: Do you record at any other studios around town?

RP: Yes, I do. In fact, I've cut a few tracks this year at other studios. Either because my studio's been booked, or just because I believe that every now and then it's good to get out and try other facilities, just to make sure that we're up to date with everything. I have done more work probably this year in outside studios than ever before, although certainly Studio 55 is my home base, and I feel most comfortable working there.

I've used Record Plant, Lion's Share — those I think are the main studios I've used this year. Last year I did some work at Music Grinder.

R-e/p: But you always try to come back and finish the project at Studio 55?

RP: Yes. Unless there was like a *very* specific reason why I couldn't work at 55. It's not so much because it's my stu-

dio; the results are as good as any place I've worked at. Everything I want is here, so there's no reason to go out. But I'm never close-minded to it; if for any reason some other studio had a particular sound that would be more right for a particular project, then I'd be crazy not to use that facility.

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

RP: I've had virtually no experience with digital, and I'm not really sold on the technology yet. I feel that there is something intrinsically unmusical about digital; until now I've been an analog man all the way. Even with digital synthesizers, there are ones that sound great for certain things, but I just *love* the good old analog sound. I think that tape hiss adds a certain "ambience" to a record, if you will, that never bothered me. I don't think anyone can say that whether a record is a hit or not depends on whether it was done digital or non-digital.

R-e/p: Several digital advocates claim that the 32-track capability of certain multitracks can be a great advantage when you start running out of tracks. Plus the ability to clone tracks with no loss in quality.

RP: I think that the operational convenience is clearly there. But for 46-track working, we've got our Studer machines and the [Audio Kinetics] Q.Lock. It can be a bit of a pain waiting for the machines to sync up every time, even though it's pretty fast. But usually you only have to deal with it in the mixing, because you make slave tapes for overdub sessions. It's not *too* much of an inconvenience, but it certainly would be nice not to have to deal with that at all. And it's great to be able to make copies with no generation loss.

If and when I make that move, it's important that it be the "Rolls Royce" of

"Richard has very good insight into lyrical content, melodies and structure — he's a really good song finder. He's been known to toss a song back to the writer, especially for the lyric; he is very sticky about that. Once in a while he will run into one of my songs that is pretty much complete in his eyes, but usually, there is a bit of work to be done somewhere before he is satisfied with it."

— **Greg Phillinganes**, solo artist and musician/arranger, currently at work on a new solo album, *Pulse*, for Richard Perry's Planet Records.

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digital machines. Otherwise, two or three years from now I may be stuck with a piece of equipment that has already outdated itself. Whereas my present system of hooking up two 24-tracks, and using slave tapes, makes a lot more sense, and has worked very well for us. Most people are still working with that system, because it saves you playing the master tape over and over again during overdubs, and losing the top end.

Even though 32 tracks or whatever *should* be more than enough for anybody to deal with, the idea of having slaves can be a tremendous working tool. Say, for instance, that you do a whole series of vocals or overdubs and use up all your tracks, and then maybe you want to come back in and do another series. You can make another slave to open up a new group of tracks, and then decide which are your better ones; you can potentially give yourself an unlimited number of tracks to work with.

R-e/p: Have you plans to release any of Planet Records product on Compact Disc?

RP: The first one we're going to do is the *Break Out* album. RCA also plans to release a CD of the first album I did with Harry Nilsson, *Nilsson Schmilsson*, which I'm kind of excited about because it was a particularly good-sounding record. I can't wait to hear the *Break Out* album on CD.

R-e/p: How do you perceive Planet Records? Does it function more as a vehicle for your artists and productions?

RP: Planet Records has been in business now for six years, which is quite an achievement in itself! Custom labels really don't work in the business anymore, for a variety of reasons, but now there are very few artists on the label; at one point I had up to 10 artists signed to the label. Right now I'm very happy just to be concentrating on the Pointer Sisters and Greg Phillinganes — it's mainly a vehicle for my productions. I'm always open to new artists coming on the label, with somebody else producing, but it would have to be of such a *special* quality for me to put my full support behind it; so far nothing has crossed my path that has inspired me on that level.

You could say that owning your own label is the highest level of a production deal that anyone could get. During the period around '76, '77, I was experiencing an increased frustration over the inability of a few record companies I was doing work for, to help my records achieve what I felt were potentially their ultimate goals. And the frustration became so great that it was actually taking some of my desire and heart out of the business for me, and I didn't want anything to dampen my enthusiasm for making records.

R-e/p: Was your frustration related to the lack of marketing of your product?

RP: Exactly. Marketing, promotion, the whole coordination of all those events. I always tended to follow my records

beyond the delivery to the label, which is when technically a producer's job ends, and I would frequently get remarks like, "We don't tell you how to *make* the record; don't tell us how to *sell* it!" So really I started Planet at that late point in my career as an outgrowth of that increasing frustration I was feeling. Now it clearly had became my job to follow the record through, check on how everything is going, and to work with the people whose job it still is to market and promote it, but at least now I had a much closer connection with that phase of the record's life.

There have been, like I say, some unbelievable elements to deal with that take away from the energy and time one is able to spend in the studio. That's the main reason why I've cut back on the number of artists, because I couldn't, in all good conscience, be president of the label and do that job for those artists and, at the same time, do what I do best: in the studio making records.

R-e/p: Do Music Videos figure in Planet's future plans?

RP: Most definitely. In fact, it's not only a role that I forsee, but a role I'm very heavily involved with. I've already done two videos with the Pointers. The first one was *I'm So Excited*, which was done almost two years ago. That video I co-directed with Kenny Ortega, who I had worked with for years as a choreographer for the Pointers. He'd never directed a video at the time, either, so we did it together, but essentially it was my concept and I edited the video together with one of the top editors in the business. Just recently I directed the "Jump (For My Love)" video, which was put together in less than a week, on a very minimal budget.

Within the last couple of days we've had an offer to do a 40-minute commercial videocassette with the Pointer Sisters to coincide with the release of the next album, which is precisely the type of thing that I've been waiting to do for some time. As far back as 1973, after I had achieved what I felt was almost as high a level of success as a record producer could hope for, I started developing a number of different film projects that I planned to direct. Once I started Planet, however, it became impossible for me to give up a year to two years of my time doing nothing but planning on directing a movie, which was okay with me because at that point I felt there was still plenty of time to do that.

But, having been very involved with not only developing a couple of film projects, but also a TV mini-series — all of which were very heavily musically oriented projects — I started looking for a video project that has the potential for a lot more depth and substance, and a high level of repeatability. That's what I'm working towards, and I think it's a tremendous transitional vehicle between records and films. Because, let's face it, the commitment to drop everything and direct a film is substantial, but I feel very confident in my ability to do that.

Not every producer could make that transition to film directing, but I think that the background and experience you achieve in making records, and working with different types of artists, will give me a tremendous advantage in making films. There's still a lot that I have to learn, but I look forward to that challenge and learning process with great anticipation.

I've learned a tremendous amount about the whole process of making videos, which again is something that came very natural to me. So I really look forward to continuing in this direction, and doing some long-form video projects; and then, within the next few years, to launch my first film. ■■■

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A Regional Sound-System Design and Rental Firm with Cost-Effective Approaches to Concert Sound Operation

by David Scheirman

The concert sound system rental business traditionally has been a relatively conservative industry. Major sound companies are reluctant to make hardware changes in arena-sound systems until the new gear has been road-tested and found to be a worthy investment. And the critical environment of a national tour with a well-known musical group is not usually selected as the proving ground for new technologies in live-performance sound system hardware and packaging; failures are both costly and embarrassing. The major PA companies have attained their current positions by finding out what *does* work, and developing cost-effective ways of offering consistent concert sound to their clients in venues across the nation.

In the ever-changing entertainment audio business, one constant question in sound-system design for traveling use is more important than ever: How much sound can be carried in how little truck space for the least amount of capital expenditure? When practical considerations, such as reliability, and subjective considerations, such as sound quality, are taken into account, in the past this rather complex equation has spawned some interesting concert sound systems.

The large concert-sound companies often find it difficult to respond quickly to changing trends in hardware and packaging. A single arena-sound system represents a significant capital investment. Changing system components, be they connectors or consoles, require time and money. And those companies that possess up to a dozen such systems face a formidable task to retrofit existing systems in a standardized fashion.

Oftentimes, innovative ideas in system design come from the smaller, aggressive regional companies that are trying to capture a share of the touring-sound market. Some such companies try to parlay gimmicks and publicity into dollars and cents, while others combine market-responsive new technology and good business practices into a steady growth rate. This article is the first of a series of regional sound company profiles, in which we will take a look at how various firms are approaching concert-sound system design for the Eighties.

From Humble Origins

In addition to being a popular concept in the live-sound business this decade, Modular Sound Reinforcement is the name of a regional sound system design and rental firm located in Trenton, New Jersey. Modular is a division of Joe's Amalgamated Industries, a corporation founded in June, 1977, that grew out of the partnership formed by friends who became interested in sound-system design and rental while attending college in Ithaca, New York. Modular Sound Reinforcement serves as a proving ground for ideas in enclosure design developed by Joe's Sound & Salami Co. The fabrication and marketing division, in turn, offers a capital base for research and development by Modular.

To investigate the company's claim that it is able to offer touring musical groups more "sound per pound," this writer took a look at two events that were held in the company's geographical region: an outdoor Bluegrass Festival on Staten Island near New York City, and a three-act show at a new-music nightclub in Princeton, New Jersey.

There are several aspects of particular interest with Modular's systems: a mechanical method for aligning the time/phase relationships of low-frequency and high-frequency components; the use of non-slant stage monitor speaker enclosures; experimentation with "wood-chemical" materials for loudspeaker enclosure construction; and the ownership of manufacturing rights for the 2005 A.D. brand of mixing consoles (miniature, high-quality boards designed to military specifications). When these technological ideas are coupled with the fact that Modular has an "apprentice" program for road engineers, as well as operating a speaker-enclosure fabrication plant and a recording studio, a very credible sound company profile emerges.

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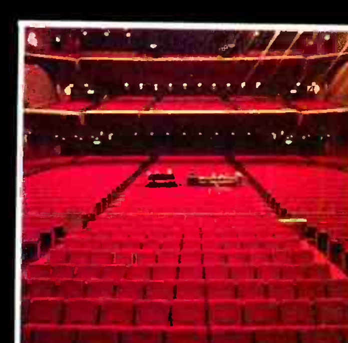

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People behind the Company

Robyn Gately (pictured right) grew up in an audio-oriented environment; his father and brother are both active in the field. Having mixed Bruce Springsteen's live show at an early point in that artist's career, Gately gained some insight into what was required of a concert-sound company that wanted to make a name for itself.

"We moved to Trenton because of its central location in the Northeast corridor," he notes. "New York, Philadelphia, Boston and Washington are all within reach. This part of the state offers good interstate access to the major population centers in this part of the country."

Gately and his partners became interested in loudspeaker-enclosure design for touring use, and parlayed that interest into a manufacturing company that builds products for nationwide distribution. "The reason we got into building enclosures in the first place was that it seemed that no one was addressing the truck-space and weight problems of touring sound," Gately recalls. "In this industry, a lot of people seem to have no view into the future . . . they are only thinking about a particular tour. However, Modular is tied to two seemingly unrelated industries: sailboat design and fusion research.

"From our interest in sailboats, we have learned things about fluid dynamics that are applicable to loudspeaker-cabinet design. The way that the chine of a racing yacht is designed will influence its speed through the water; the Australians proved that in the last America's Cup Race. In like manner, the design of an enclosure will determine the 'speed' of the sound waves generated at that enclosure. I am not talking about the speed of sound, but rather the amount of turbulence generated in the air at the baffle-board."

Diverse Design Influences

Gately also feels that the fact that one of the company's consultants works on the Tokamak fusion research reactor at Princeton gives the company an edge in the development of new technologies. "The *real* electronics industry, the cutting edge, is *not* the audio industry," he offers. "The things that are seemingly roadblocks to the development of more powerful and efficient amplifiers and transducers are being broken down with recent developments in the physics of wave travel and pulse-trains. There have been new developments in other industries that are staring the concert sound business in the face, and no one seems to be addressing them.

"Aerospace is one example. Today, a section of floor panel for a new airliner weighs only 1/100th of what the same-sized panel weighed on an old DC-3. And yet, the typical large loudspeaker cabinet for sound reinforcement use is probably *heavier* than those in use 20 years ago!

"Not only is there much room for new materials research, but the airline industry may have some things to teach us about how we can improve transducer performance with smoke-testing in wind tunnels."

The engineer offers that comparison of a Modular speaker stack with the same number of cones and drivers in a typical composite box used in the concert-sound industry today would show their boxes to be a measurable 3 dB hotter. "That's with the same input signal, the same transducers, and without the use of horn-loading on the cones," he claims. "Horn loading would be one way to get more sound out front, but it is not the *only* way. And, in our opinion, it is not the best-sounding way. Developing a cabinet that is more rigid, so that acoustical energy is not lost to mechanical energy that actually vibrates the box . . . that is what it is all about."

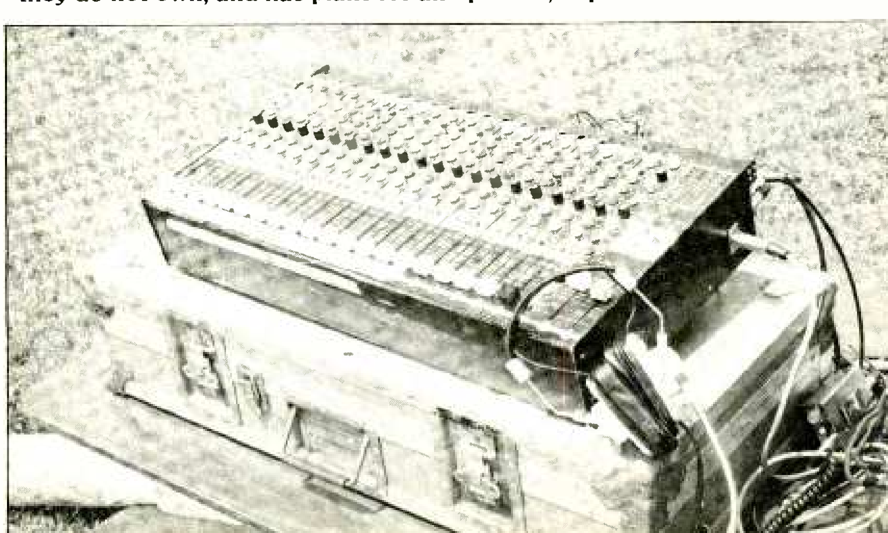
He looks forward to the development of hanging sound systems that will lead to a reduction of reverberation in arenas considered to have acoustical "problems," due to the use of smaller arrays designed with the idea of a single point-source in mind. "The smaller the actual system components, the better shape you are in for sight lines," Gately says. "And flying the system directly over the performance area will help with time differentials."

Studio as Classroom

Modular's Vann K. Weller (pictured left) oversees operation of the company's recording studio, among other responsibilities. "Actually having the controlled environment of the studio for testing new gear has been very beneficial to the development of our concert sound systems," he notes. "Not only do we have an acoustically treated room for testing loudspeaker enclosures, but it also serves as an excellent 'classroom' for teaching the participants in our apprentice program many basic ideas about mixing, as well as the use of things like compressors and equalizers. And, the studio is very valuable for new artist contact. The fact that we also use it primarily to generate income by doing sessions for clients helps a lot at times of the year when sound system rental is not so active."

The company places occasional advertisements in local newspapers, Weller says, for apprentice helpers on

Figure 1: Modular Sound systems use 2005 A.D. mixing consoles, a unique compact package built in limited quantities approximately 10 years ago to military specifications. The company services those remaining consoles that they do not own, and has plans for an updated, expanded version.



SOUND COMPANY PROFILE
Modular Sound Reinforcement

sound system rental dates. "Probably only one guy out of 30 who answers the ads will end up actually having what it takes to do the job. We hold regular evening classes for the apprentices. Those who get through the whole learning program, and still show an interest in actually doing concert sound work, we offer employment to after a complete evaluation of their capabilities."

Weller notes that the differences between mixing techniques used in the studio and in live performance have been narrowing over the years. "In our apprentice program, we stress the fact that the contemporary sound systems on the road are *much* more sophisticated than they were even a few years ago. The computer-controlled devices, the special effects processing; you are seeing it all out on the road now with the big systems. The person just entering this business will have a much more versatile array of job skills if he or she is equally at home mixing music for tape and for an auditorium."

Joe's Recording Studio currently offers 16-track capability, and is built into the same old Gothic-style build-

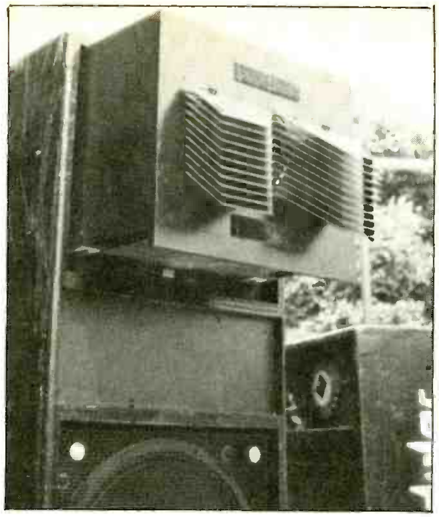


Figure 2: The "Atomic Pile" speaker cabinet features Slide-Align technology. A mechanical method of adjusting the time/ phase relationships between the LF and HF components precludes the use of electronic signal processing.

ing (a former funeral parlor) that houses Modular Sound Reinforcement's offices, electronics design lab, and system warehouse. Site improvements include a sand-filled drum riser, fiberglas-insulated burlap-covered walls, and a mahogany-panelled control room. "The shape was designed to eliminate the short reflections that cause comb filtering, while retaining the longer reflections that give the room a sense of depth," Weller offers.

The custom Data-Mix 28/24 console was formerly installed at the Hit Factory and the Sound Palace; tape machines include a 3M M79 16-track two-inch, a Tascam Series 70 eight-track half-inch, and a Scully two-track.

Compact System Design

George "Dewey" Williamson (pictured right) works with total sound-system design and packaging, as well as overseeing the labor for the loudspeaker enclosure manufacturing facility that employs 15 people.

"One of the most important things to us is that the total system must take up as little space in the truck as possible," he explains. "Some companies seem to work on making one part of the system small, while overlooking other things such as huge, 'dinosaur' mixing boards.

"One of the things that appealed to us when we first took over the servicing of the 2005 A.D. boards was the small size. When they were first built, they offered probably 10 dB better signal-to-noise ratio than any commercially-available consoles. They are built to military specifications, with things like gold-plated edge-connectors. They are extremely clean, extremely simple, and sound extremely good. We are working right now on our first prototype 25/8 monitor-mixing console. When we took over the 2005 A.D. consoles from the designers, there were enough parts for another 25 boards or so. Right now, probably all of the first

Figure 3: The rear of an Atomic Pile, showing the JBL 4425 HF compression driver in fully-extended mode on spring-suspension travelling rails.

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generation that were built between 1974 and 1980 are still in existence. These include consoles intended for live PA and remote recording use that came in 17-and 25-channel versions; the odd input format was chosen to include spare modules. Jerry Fat is working on new companion circuitry for the 25×8 prototype which may include VCAs [Figure 1]."

Williamson details some of the system gain-stage concepts used when packaging a Modular system: "We have a patchpoint on the console that is post-mix, pre-master fader. The line goes right to the limiter and EQ unit for the left and right outputs, and then back into the console. That way, the console meters show post-limiter and EQ levels. So, when your board meter hits the red, your power amps theoretically hit the maximum power that you want them to pass signal on to the loudspeaker system.

"For example, on a Phase Linear 700, rated at about 300 watts apiece, feeding two 15-inch speakers per channel, you are pretty close to the rated power capability of the speakers. So you want to calibrate the system so that when it hits zero VU, the board is hitting zero VU. In that way the console meters show you the *actual* output gain of your whole system electronics."

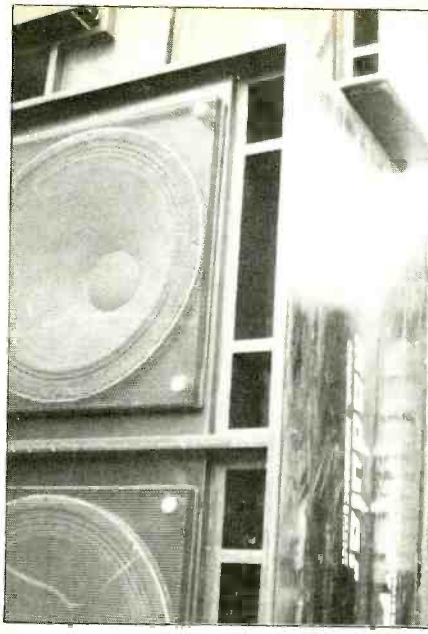

Figure 4: The "Multiple Warhead" enclosure, loaded with JBL K151 18-inch drivers, is constructed of a wood-chemical hybrid material that reportedly offers a high rigidity factor.

Modular sound systems currently are supplied with dbx Model 118 limiters, Klark-Teknik graphic equalizers, and DeltaLab DL4 delay units. UREI Model 525 electronic crossovers split the signal.

"In our apprentice program, we try to stress that an equalizer is there to be used only if necessary," Williamson explains. "It is there to correct for acoustical room deficiencies, *not* to make up for problems that are built into the system. We try to get rid of those problems at the *source*."

John Fish (pictured left) has contributed many innovative ideas to

Modular's sound systems, particularly in hardware application and shop efficiency. As shop foreman, he has developed a computer program for use in determining wood cuts for plywood sheets, as well as unique solutions to Modular's hardware requirements for the Slide-Align™ cabinets.

"Since we have to cut up a lot of plywood for our regular speaker boxes, it takes up a lot of time to figure out what cuts to make on each sheet. So, I worked up a program to tell us just what cuts to make on what sheets of wood when we are building a lot of boxes. It takes just a couple of minutes to load the program. Then, I can go on to something else while the computer figures out things for me. Instead of just doing basic math with pencil and paper, I let the computer work it out for me — and that leaves time to work on other things!"

The company's Slide-Align concept uses sliding suspension rails for the movement of the high-frequency horn and driver in and out of the front baffle-board of the cabinet, thus offering the ability to physically change placement of the horn and its relationship to the low-frequency component. Cabinets using this device are given such names as "Atomic Pile" (Figures 2 and 3). A companion low-frequency, double 18-inch enclosure is known as a "Multiple Warhead" (Figure 4).

"To come up with a hardware solution to a design concept, I usually just cruise the aisles of the local building-supply store, and try to imagine how other people have solved problems in a different type of industrial application," explains Fish. "One place lets me look around in their back shelves, and humors me . . . sometimes I find a 30-year old piece of hardware that gives me an idea for a new use. Removable wheels, for instance . . . wheels really bother me on low-frequency cabinets; they rattle and cause all kinds of problems. Locking

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A small universe

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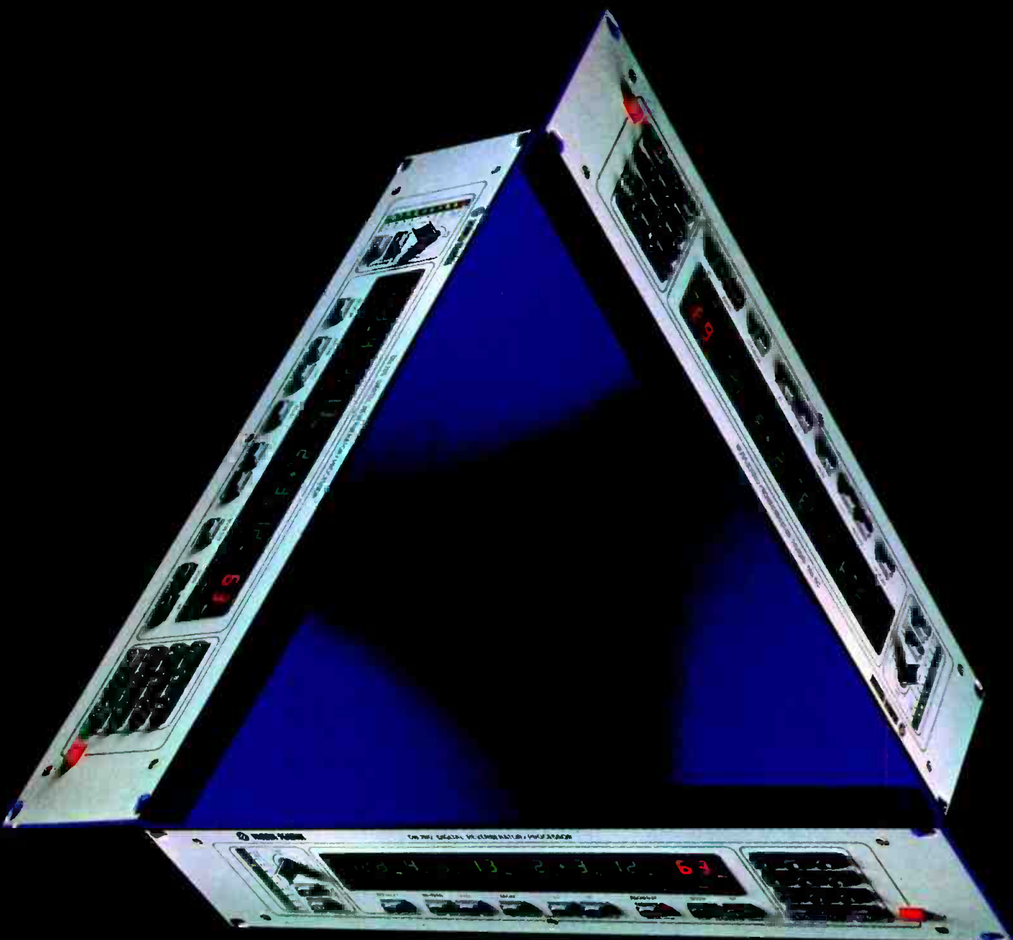

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. . . continued from page 60 —

wheels don't really work well. A locking-track idea came to me that way."

Modular's stage monitors differ from most in use today in that they are not of the slanted variety. "What we did was to try to develop a great-sounding monitor, which turned out to be rectangular," Fish comments. "Then, we approached the problem of how to tilt it up for the performer, with varying selectable degrees of tilt: a locking pin with different holes for different positions is what I came up with. The rectangular boxes take up less room than slanted monitors. But we did it mostly for the way that they sound [Figure 5]."

Live Performance

To get a feel for what Modular Sound Reinforcement's systems actually sounded like, I witnessed two live-performance events in June of 1984. The company's technicians used basically the same type of system to cover a 300-person capacity, new-music nightclub, and an outdoor field covering approximately 20 acres that hosted a bluegrass festival on double stages.

Figure 5: Modular's stage monitor cabinets consist of a two-way non-slant box housing a 12-inch speaker and a one-inch driver. An additional auxiliary HF section is removable.

A typical Modular system features 2005 A.D. mixing consoles for both house mix and stage monitors; two of the consoles are easily strapped together for events requiring greater input capabilities. UREI Model 525 crossovers are set at Crossover points of 280 Hz and 1.8 kHz for a three-way system.

Transportation requirements for the systems were minimal, as electronics and amplifier racks were kept small and the entire system was quite spartan. The small, lightweight Carver PM1.5 amplifiers were used to power all functions except the low-frequency bandwidth in the main system. Here, Phase Linear D-500s were employed (Figure 6).

Microphone selection available at each event included Shure SM-57s for vocals, AKG D12Es for kick drum, Sennheiser MD-421s for other percussion applications, and Beyer M400s. Audio-Technica ATM-10 condenser and Sony ECM-150s were also available, along with a complement of direct boxes.

The company's unique loudspeaker systems comprised three types of cabinets that utilize JBL cone speakers, horns and compression drivers (Figure 7).

"Before settling on speakers, horns and drivers, we tried just about everything that is commonly available," notes Robyn Gately. "Nearly everything that JBL, Electro-Voice, Renkus-Heinz and Community Light & Sound makes, we tried. We took each version of the system out to at least a half-dozen gigs for trial purposes in the field. The JBL 2307 horns with the detachable 2308 lens units were chosen due to their dual coverage pattern, and low throat distortion characteristics."

The company's low-frequency enclosures, known as "Atomic Bombs" and "Multiple Warheads," house either one or two 18-inch JBL K151 loudspeakers in vented, front-mounted enclosures. According to Gately, the front "lip" around the edge of the baffle-board is an important part of the enclosure's performance. "We've learned that from the racing yacht industry," he explains. "The more you round off the chine of a boat, the less effective it becomes in moving through the water at high speed. Even using a 3/16-inch router to give the edge a slight curve instead of a hard-angled edge will give a noticeable deterioration in performance." Gately claims to be able to draw a parallel to this phenomenon in loudspeaker enclosure design.

Modular's two-way cabinets come in two versions: the Atomic Pile houses two JBL K120 12-inch speakers mounted in a vented enclosure, as well as two Model 2425 one-inch compression drivers. Another version houses a single Model 2441 two-inch driver. The "Subatomic Pile" contains a single 12-inch speaker with one Model 2425 compression driver.

Figure 6: Modular's power amplifier racks house Carver PM1.5s for mid and high amplification. Phase Linear D-500s power the LF components.

"Originally, we tried five-inch speakers in the Piles, six per cabinet, and let them handle the 1 kHz to 5.2 kHz bandwidth. However, the available speakers couldn't take the power required for the 5s to compete with the other components. It has turned out better as a three-way system using the 12s.

"And these systems are essentially flat . . . we try not to use EQ unless absolutely necessary. We've found that almost every EQ unit currently being offered on the market will change the sound of the system just by being inserted in the signal path, even when it is set on 'flat.' And we try to train the guys to not depend on our real-time analyzers . . . in fact we don't even allow them to use them at shows. The human ear is the *best* test instrument."

Indoor Systems

An L-shaped crowded nightclub — the Tin Lizzie Garage in Princeton, New Jersey — was under audio assault by several local bands when this writer arrived to hear the Modular sound system. One performer described his group's music to me as "Punk Fusion." With a General Instruments sound-pressure level meter, I measured average program material at 115 dB, with 121 dB peaks at the console (located approximately 30 feet in front of the small performance stage).

Figure 7: A typical MSR speaker stack: a double-18 "Multiple Warhead" cabinet supports two "Atomic Pile" two-way boxes. The smaller cabinet is known as a "Subatomic Pile." The entire sound system, including mikes, stands, cables and electronics, was trucked to the site in a one-ton van.

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One Atomic Pile, one Subatomic Pile, and three Atomic Bombs were supplied in each of the two speaker stacks that flanked the stage. Low-frequency components were stacked directly on the floor, but the club's low ceiling offered few other choices.

The speaker system was more than adequate to place the mixed program material above the ambient noise in the room (which included that noise issuing from the stage). Due to the room's design, full-frequency reinforcement of the music to all areas of the club was not possible. On-axis, the stacks sounded clean and full.

> **Author's Note:** Any specific mention in this article of manufactured audio equipment is not to be taken as an endorsement, implied or otherwise. The arbitrary choice of this growing sound reinforcement firms was made with reader interest and education as the sole factor in mind — **DS**.

Outdoor Systems

Snug Harbor on Staten Island is a restored sailors' rest home from the last century. The park-like outdoor area makes an ideal setting for civic affairs and live performance events. Modular recently provided two separate sound reinforcement systems at this venue for the New York City Bluegrass Festival, a yearly event. Crowds numbering up to 4,000 persons heard a lively sampling of nationally-known acoustic and electric acts, as well as witnessing a contest held for local groups.

Besides serving as the main performance area, the 40-foot Festival stage supported Modular's speaker stacks. Two Multiple Warheads, two Atomic Piles and one Subatomic Pile were set up per side, for a total of four, 18-inch cones, six, 12-inch cones and eight compression drivers.

With the bluegrass music as program material, an average SPL of 92 dB was maintained at the house console, located 100 feet from the stage. The Modular stacks offered consistent, even coverage. Full-frequency reinforced sound was available up to approximately 400 feet from the loudspeaker locations. The speaker stacks seemed to be fairly directional in the outdoor setting, even at lower frequencies.

* * *

For a regional sound company to parlay local successes into a nationwide business, hard-won capital must be poured into research and development as well as new equipment acquisitions. Modular's rental sales manager, Paul Collins (pictured right), notes that seeking work with new up-and-coming acts, pushing into a variety of marketplaces, offering monitors-only system rentals, and providing compact, full-sound systems that take up less truck space, are all a part of the company's marketing plan for future growth.

Documentation of some of the company's claims concerning enclosure performance would certainly help to bring those claims into sharper focus. Accepted standard test procedures with calibrated equipment help to more clearly determine whether or not specific enclosure designs warrant greater levels of production.*

I did find this group of young sound system technicians to be seemingly devoted to higher-quality concert sound, and noted with interest their overall grasp of the many facets of that business. It is from such beginnings as this that the major sound reinforcement firms of the next decade will spring. ■■■

*According to Robyn Gately, to help remedy this situation, the company recently purchased an Apple IIe computer with the IQS FFT Time-Spectrum Analyzer program — *Editor*.

The New Bose® Pro.

A Hands-On Commitment to Professional Sound.

by John Geheran

Vice President, Sales and Marketing

We recently took a critical look at Bose Pro—and found a need for changes. We've always had excellent research, development, production, and service facilities, but that's not enough. We must also respond to the specific needs of the marketplace. So we've begun listening to you and learning how to better use our resources to serve you.

We're committed to your success, and we want to help you grow and develop. We've installed a team of professionals that know where you've been and where you're going. In addition, we're developing a variety of new products, design and sales tools, and support programs. Finally, we're dedicating this year's advertising space to a series of articles which deliver the kind of information you can really use. But first, we'd like you to meet the new Bose Pro Team.

Paul McKinley, Marketing Manager

We needed someone with formal training in both engineering and music as well as hands-on experience in systems design and installation, live performance, and business operations. We chose Paul McKinley, previously our audio-visual manager.

Following college, Paul devoted several years to designing and installing audio, multi-media, and video systems in nightclubs, museums, and industry. In addition, Paul has 12 years' experience as a professional musician. Since joining Bose, he has contributed to a number of technical and marketing projects.

John Stiernberg, Sales Manager

We needed someone with strong organizational and managerial skills as well as practical experience. John, having been our Midwest sales rep, has strong experience as a retail salesperson, manufacturer's rep, technician, and performer.

John has recorded albums, emceed major festivals, done seminars on college campuses, and even produced music for a PBS documentary. He's also a member of NARAS and the American Federation of Musicians.

Steve Romeo, New Business Manager

We needed someone who could direct and expand the technical services necessary to support growth in the professional sound business. Steve has experience as a sound contractor, a concert engineer, and a creative problem solver.

Steve has installed sound and engineered concerts in everything from corner bars to football stadiums. He's already been involved with key Bose installations around the world, and has conducted numerous seminars with contractors and consultants.

What can you expect from the New Bose Pro Team? First, expect to see them in the field. Second, expect useful information. Finally, expect the quality products and services you deserve.

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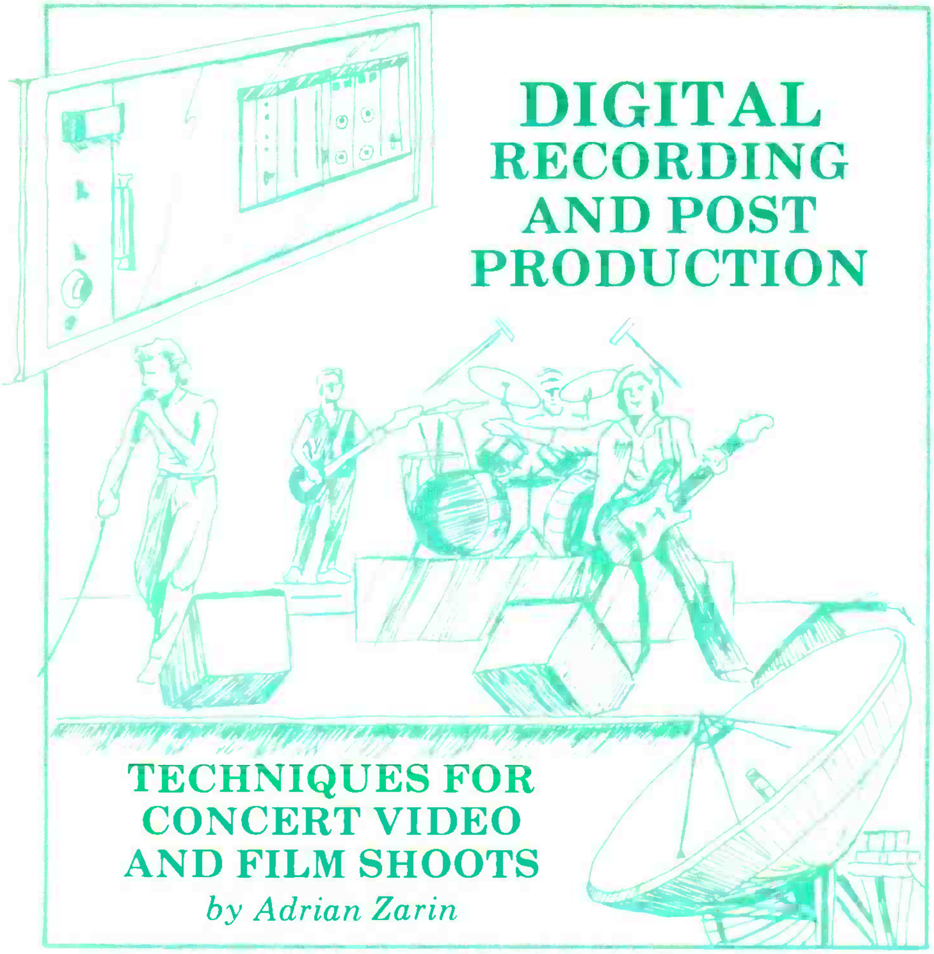

DIGITAL RECORDING AND POST PRODUCTION

TECHNIQUES FOR CONCERT VIDEO AND FILM SHOOTS

by Adrian Zarin

There is every indication that the marriage between digital audio and visual programming will be as happy and fruitful as it is inevitable. As *the* budding sound technology of the decade, digital audio is a natural and logical consort for the visual media, also destined to reach new heights of prominence thanks to the dual impact of Music Video and Stereo Television.

Two recent groundbreaking concert productions — the Talking Heads' feature film, *Stop Making Sense*, and the Showtime simulcast special, *Stevie Wonder Comes Home* — demonstrate that the union of digitally recorded music with live performance footage is getting off to an excellent start. Although each show was created for a different medium, the makers of both presentations faced similar problems at crucial stages of production — particularly in the areas of synchronizing digital audio with video and film equipment, and overcoming the inherent limitations of the final playback format. The trails blazed by both these productions in solving such problems no doubt will provide useful models for future productions of this nature.

Stop Making Sense, the Talking Heads' concert feature, documents the group's three-night stint at Hollywood's Pantages Theatre last January. Initially recorded analog, audio for the show was transferred to digital for editing, overdubs, sweetening and subsequent mixing. Providing the digital know-how for the project was John Moran, president of Digital Services, Inc., based in Houston. Talking Heads leader David Byrne, and drummer Chris Frantz, played an active role throughout show's post production, working with Moran, producer Gary Goetzman, director John Demme, music editor Jim Pryor, and mixers Joel Moss and Steve Maslow. Premiering at the San Francisco International Film Festival earlier this year, the film garnered praise for its high-quality audio and exciting visuals. The movie is slated for future commercial distribution.

A multiple-camera film shoot captured the action at the Pantages, while a Record Plant/L.A. mobile truck made a 24-track analog recording of each night's performance. To ensure synchronization, film cameras and tape machines were fed SMPTE timecode from a common source, and which become the basis for the subsequent integration of digital and analog tape machines with film and video equipment during post production. (In the case of the film cameras, video monitors located near every camera position enabled timecode displays to be captured on film at the start and end of each reel.)

The remix and sweetening process began at Oceanway Recorders in Los Angeles, where the analog tracks were transferred to digital and conformed to the edited picture. A synchronized system of four audio and video machines was set up to enable the audio crew to transfer and conform the music tracks to continually updated rough picture edits furnished on videotape by the show's picture editors.

Analog/Digital Hybrid

A maximum of 22 original analog tracks recorded on an Ampex ATR-124 24-track were transferred to Sony PCM-3324 digital multitrack, while the picture corresponding to each musical segment was played back on a Sony BVU-800 U-Matic. Audio and video transports were locked up via an Audio Kinetics Q.Lock 3.10 SMPTE synchronizer. A Sony PCM-1610 digital two-track processor bearing master timecode acted as a master clock to the Q.Lock and BVU-800, while also bearing the composite stereo mix to be laid back to the videotape for reference.

Having been shot on film, edited on videotape, and then transferred back to film as the final release format, the project presented its share of synchronization problems, as John Moran explains: "One of the major considerations that the music people had was, 'If we're doing all of our work in sync to video, how do we know that it's going to go back and sync up to film? My simple plan was to use the two sampling rates of the 1610 — which correspond to the different frame rates for video and film — to drive the video machine and then, later on, the film chain. It worked out fine.

"What we did was comp sync the video by using the 1610 to jam the BVU-800, and bring it up to film speed. This was acceptable because we eventually had to end up on film; video was purely a reference.

"Before we started dubbing from the analog to the digital multitrack," Moran continues, "we went through and pre-striped all the digital tapes with the final timecode that we wanted to be our master code on the Sony 1610. We could then figure in the offset that we needed to make the original timecode on the analog multitrack master match up with our new master timecode. We did it in such a manner that the master SMPTE and the reference video we were getting would sync up right off the bat. After that, it was simply a matter of doing offsets with the ATR-124 to make sure it would fall in with the other machines."

Rather than transferring all of the audio tracks to digital, and then con-

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forming them to picture, analog tracks were transferred on an as-needed basis only, based on the picture edits that were furnished.

"We would transfer from the ATR to the Sony multitrack in sync with the video track that we needed for that particular segment of tape," explains Moran. "This way, we were virtually doing electronic editing in sync to the picture."

Mixed Timecode Sets

This transfer and editing system necessitated the use of two different sets of timecode numbers, both of which were burned into the reference video picture. One set provided an ongoing reference of consecutive addresses running from the beginning to the end of the show. The second timecode window displayed addresses for individual segments that were being integrated into the whole.

"With all three machines running in sync," says Moran, "we would roll until the SMPTE addresses in the first window indicated that we were at an edit point. At that point, we would pull off the [master] reel [of audio tape], put on the second reel with that particular SMPTE address on it, sync that up to the picture, come up to the edit point and punch in all 24 tracks on the Sony machine. In this way, we conformed the three day's worth of recording into one composite show."

For Moran, the transfer, editing and synchronization techniques used on the Talking Heads project point towards the direction that future productions of this nature will take. "David Byrne, Gary Goetzman, myself — everybody who was on this project, I believe —feels that the techniques we developed will become more and more common," he says, "because they worked *so* well.

"One of the things you can see here is that video machines and digital multitracks are pretty similar in the way that they work: they are locked to their internal control tracks. If all of these [transports] are being driven by the same device, they're *not* going to go out of sync. The drift rates we were seeing between the video and digital machines were on the order of eight bits or eight subframes per 10 minutes — a highly negligible drift rate, and one I don't believe is even possible in the analog domain."

Left to right: Joel Moss, recording engineer; Steve Maslow, music re-recording engineer; and Jim Henrickson, music editor, at Warner Hollywood's Stage A.

Clockwise from top left: music editor Jim Pryor; Jack Keller of Warner Hollywood's optical transfer department; John Moran of Digital Services; and Doc Goldstein of Warner's technical department.

Having edited and conformed the audio tracks to the edited picture, Moran and crew had 13, 10-minute reels of digital multitrack tape bearing the complete 87½-minute show. The next step was sweetening and overdubs. Some dialog changes had been necessitated by the removal of an intermission in the live show to create a continuous film performance. There were also some percussion overdubs and "fixes" to be taken care of. All of these additional tracks were recorded directly onto the digital multitrack master in sync with the picture.

Mixdown of the completed digital multitrack master began at the Goldwyn Sound Facility on the Warner Hollywood Studio complex, since Moran felt that it was imperative to bring the project into a film facility at this stage. "Because this is a movie that's going to be seen in theatres," he explains, "in order to get the sound right, we needed to mix in a theatre. If we were to mix it to video in a sound studio, the sound wouldn't come out right when you played it back in a theatre."

Which meant, of course, that the audio now had to be synchronized to film on the dubbing stage. The PCM-3324 was interfaced to the Goldwyn film chain using a BTX shadow synchronizer and a Magna-Tech Model 9F unit that converts film feet and frames information to SMPTE timecode locations, and vice-versa. The timecode track recorded on the Sony PCM-1610 again acted as master clock for the 3324 and, this time, for the film chain. By bouncing down to two open tracks on the 3324 multitrack, Moran and crew avoided the complications of synchronizing an additional tape machine for mastering.

The same basic set-up was used at Glen Glenn Sound where mixing also was carried out. "Toward the end of the project we did a lot of going back and forth between Glen Glenn and Goldwyn," Moran recalls. "Just as someone would change back and forth between monitors on a record mix. There was a lot of classical record mixing techniques in our approach. We tried to gear the mix to as many different systems as possible, which is why we went back and forth from room to room — to get an idea of what it would be like in different theatres."

Also borrowed from record-session procedures, Moran explains, was the technique used to apply overall equalization to the final mix. "At the very end, we did basically what is done in record mastering, and took one last pass through the system in the theatre, applying EQ to the final mix just as a mastering engineer might. It was very subtle equalization — nothing radical — but I don't know if anybody has done this sort of 'mastering,' if you will, on a soundtrack before."

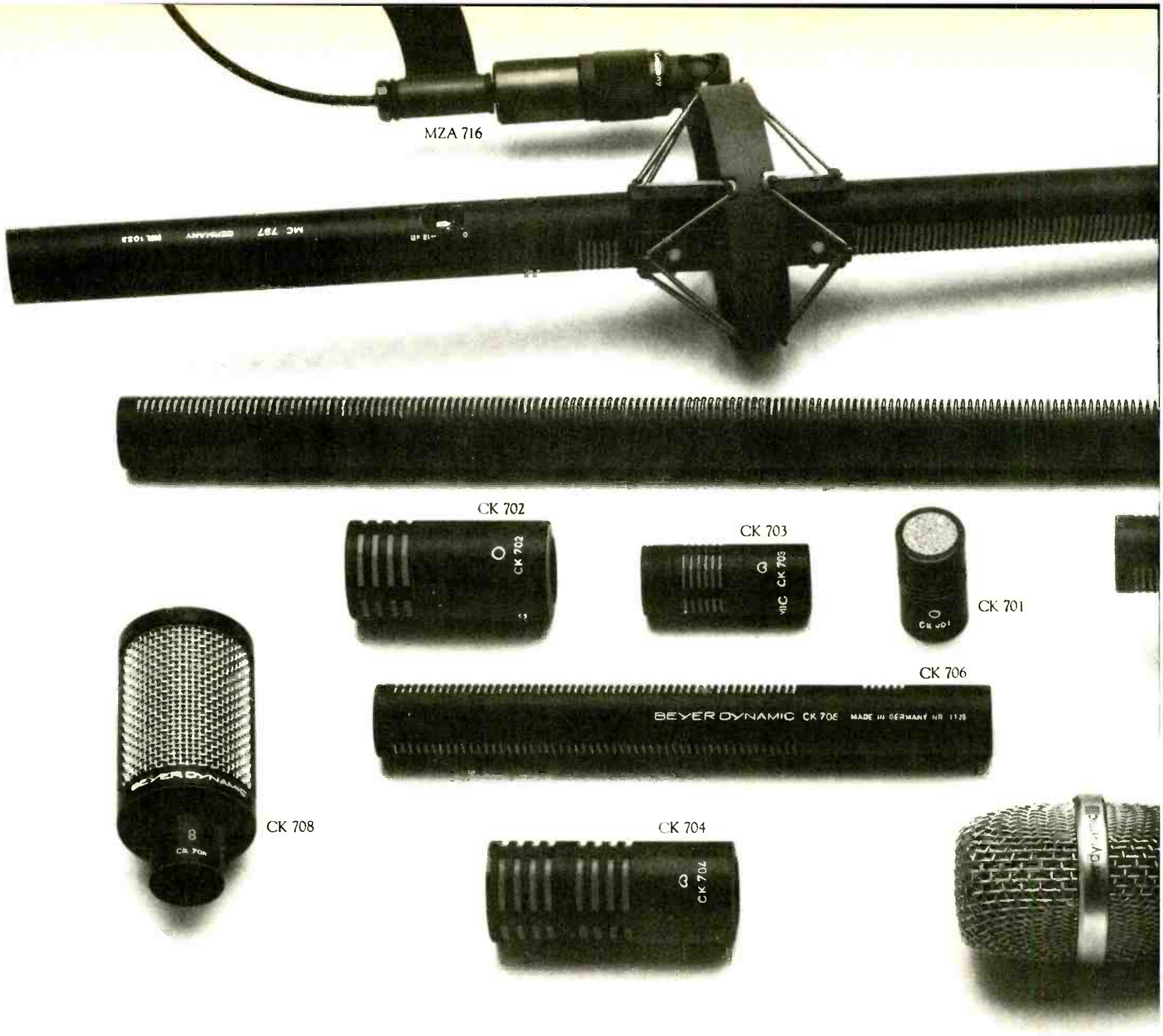
Master Stereo Mix

During this final pass, the stereo Lt-Rt mix was monitored through Dolby's standard theatre matrixing system, to check the effects of the matrix. The idea of using the Dolby Stereo system, which by definition includes A-type noise reduction, with digital audio was one that Moran came to with mixed feelings, as he explains: "Gary Goetzman says I got the damndest look on my face when he told me the project had to be Dolby-encoded. As if I were thinking, 'You're doing to Dolby my beautiful digital signals!'

"In a perfect world, where we could get digital reproduction in the theatres, the Dolby encoding [A-type noise reduction on the Lt-Rt matrixed stereo mix] wouldn't be necessary. But, right

... continued on page 76 —

Walter Gest with Sony PCM-3324 digital multitrack linked to projector and 35mm mag chain in Warner's machine room.



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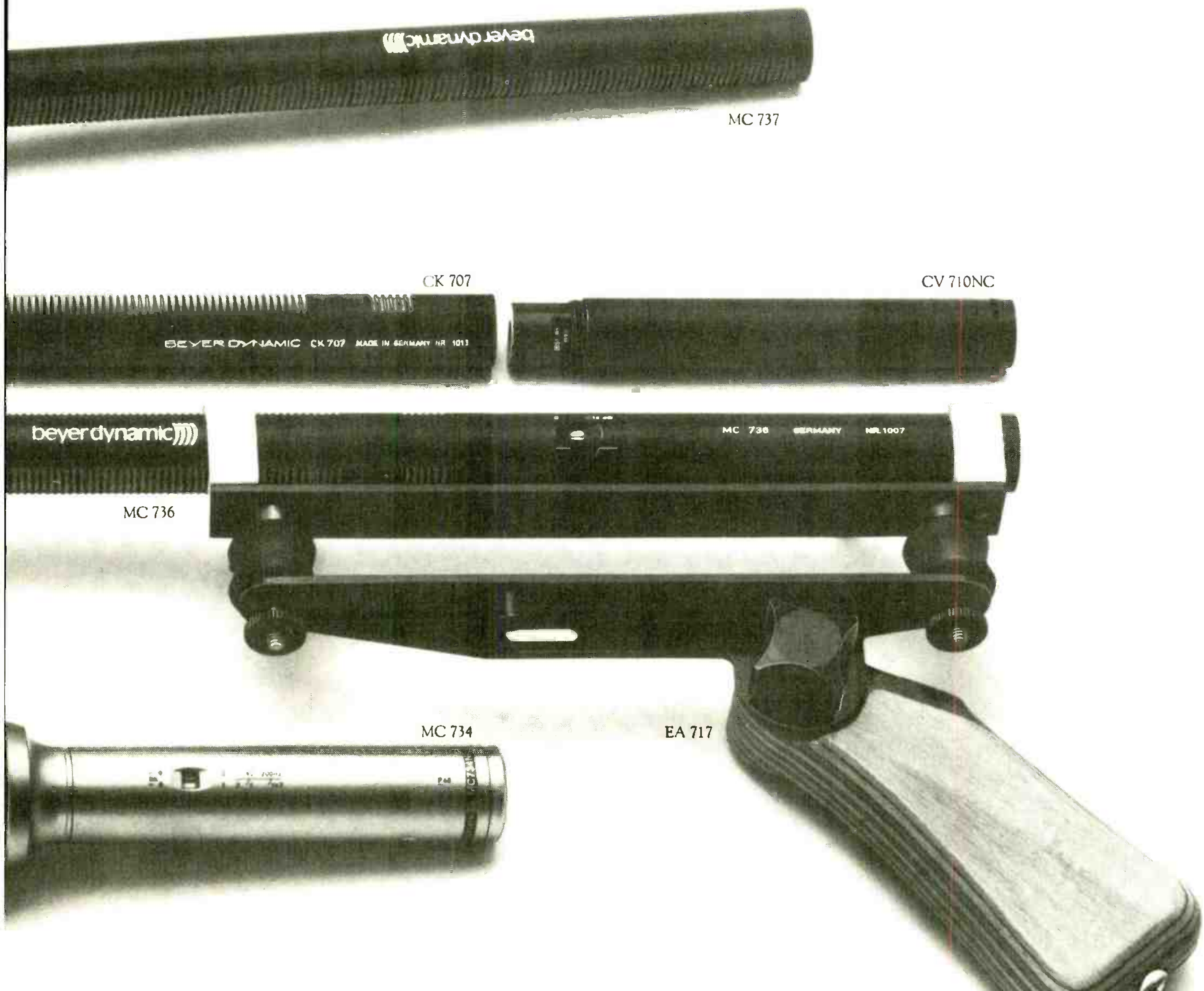
New Super High Performance Mics

The ongoing technical evolution of Beyer condensers has produced several newly designed mics offering matchless performance and long-term reliability. The new MC 737 long shotgun (lobe pattern) and MC 736 short shotgun (cardioid/lobe) utilize a built-in switchable bass rolloff to effectively suppress low end noise and rumble below 200 Hz and provide increased intelligibility for optimum speech and music recording. A 12 dB attenuator prevents high sound pressure levels from overloading the mic's internal electronics without coloration. These mics also feature an extended frequency

The Dynamic Decision



without typical condenser sound.

response of 40 Hz to 20 kHz (± 2.5 dB) and an excellent signal-to-noise ratio of 74 dB.

The new MC 734 is the only "studio" condenser mic that delivers the highest standards of performance in *any* recording or live concert situation. An extremely flat frequency response from 20 Hz to 18 kHz (± 2.5 dB) insures total accuracy while a 3-step filter compensates for proximity effect in close-miking situations. To maximize the MC 734's performance onstage, a unique stage resonance filter which reduces rumble and extraneous noise is combined with the mic's ability to withstand sound pressure levels of 138 dB (1 kHz ≦ 0.5% THD).

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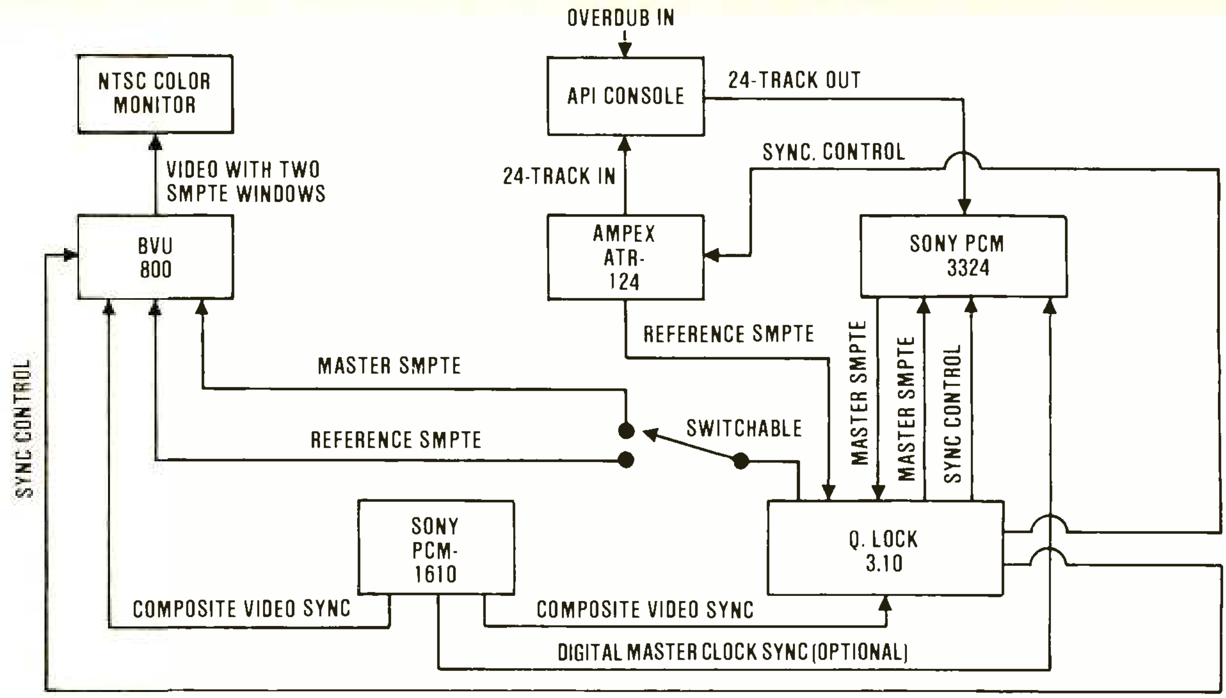


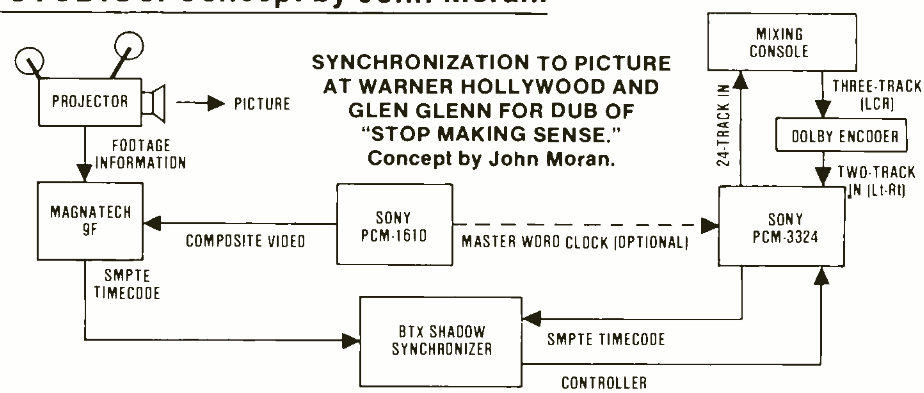
CONFORMING TRACKS TO FILM/VIDEO EDITS AND OVERDUBS AT OCEANWAY STUDIOS. Concept by John Moran.

— *continued from page 73 . . .*

now, the top theatres in the country —those equipped with 70mm projection — all feature Dolby Cinema Processors. If you're going to get the most out of these facilities, that's the way you're going to have to go; its' a fact of life."

The production also broke new ground while transferring the final soundtrack to film. Transfer of the final Dolby two-track master was made directly to optical negative from two tracks of the PCM-3324 bearing the final mix. Thus, the intermediary stages of a separate master audio tape and magnetic stock were bypassed.

"Assuming that you have no generation drop with digital," says Moran, "what we've done essentially is to go right from the original tapes recorded live at the show straight to the optical. We skipped anywhere between three and five generations.



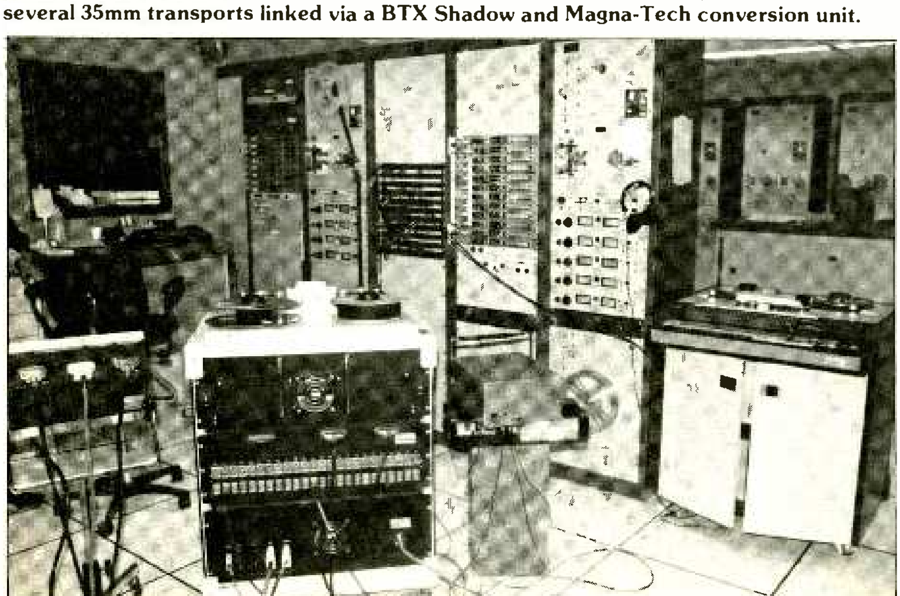
SYNCHRONIZATION TO PICTURE AT WARNER HOLLYWOOD AND GLEN GLENN FOR DUB OF "STOP MAKING SENSE." Concept by John Moran.

"When going directly from digital to optical, the only limitation you have to contend with is the bandwidth of the optical stock, which only goes up to about 14 kHz. But what you gain from the optical is an incredibly fast transient response. Because you're dealing with light, essentially, the rise time is very fast. So, in skipping those generations, we got tremendous transient response and clarity.

'The way I see it, there's no reason for people *not* to go directly from digital to optical on every digital film project from this time forward."

Looking back on the project, Moran's only regret is that the concerts were not recorded digitally to begin with. It's a situation he will be able to rectify, however, on a new project he will be undertaking with the Talking Heads and Gary Goetzman. "This will be a dedicated video project that they're tentatively planning to do in Texas," he reports. "I have not been appraised of the total concept behind the project; it's still in the planning stages. But this time, we'll be able to do it digitally *all* the way through."

Full extent of hybrid analog/digital synchronization: Sony PCM-3324, Otari MTR-90 and several 35mm transports linked via a BTX Shadow and Magna-Tech conversion unit.



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DIGITAL RECORDING AND TRANSMISSION

"STEVIE WONDER COMES HOME"

A Complete Digital Production with Video Synchronization

When it comes to video projects with fully digital audio, significant ground has already been broken with *Stevie Wonder Comes Home.* The one-hour Showtime special, culled from Wonder's recent appearance at Detroit's Masonic Temple, originally was shown on June 18 in conjunction with an FM audio simulcast. According to Lon Neumann, Wonder's long-time technical director in charge of audio for the show, Stevie wanted the show to be digital from the first. "I didn't even have to ask him," he laughs. "The attitude was: '*Of course* it's going to be digital.' To my knowledge, a fully digital project of this nature has never been done before. It took a little selling to get [producer] Ken Ehlerich and [director] Walter Miller to go for the idea. But I had Stevie behind me all the way."

Each concert from Wonder's four-night residency at the Masonic Temple was recorded using remote facilities furnished by Record Plant/New York, under the supervision of Dave Hewitt. A pair of 3M M81 Digital Mastering System 32-tracks were provided by Frank Dickenson, of Digital by Dickenson, who also played an active role throughout the project.

As Neumann explains, two multitracks were used in an overlapping mode to ensure that nothing would be lost during the concert recording. "I wanted to be *real* safe, so I started the first machine halfway through a roll of tape. I started both machines rolling at the same time; they were overlapped. One machine would be in the middle of a roll when the other machine got to the end of a roll.

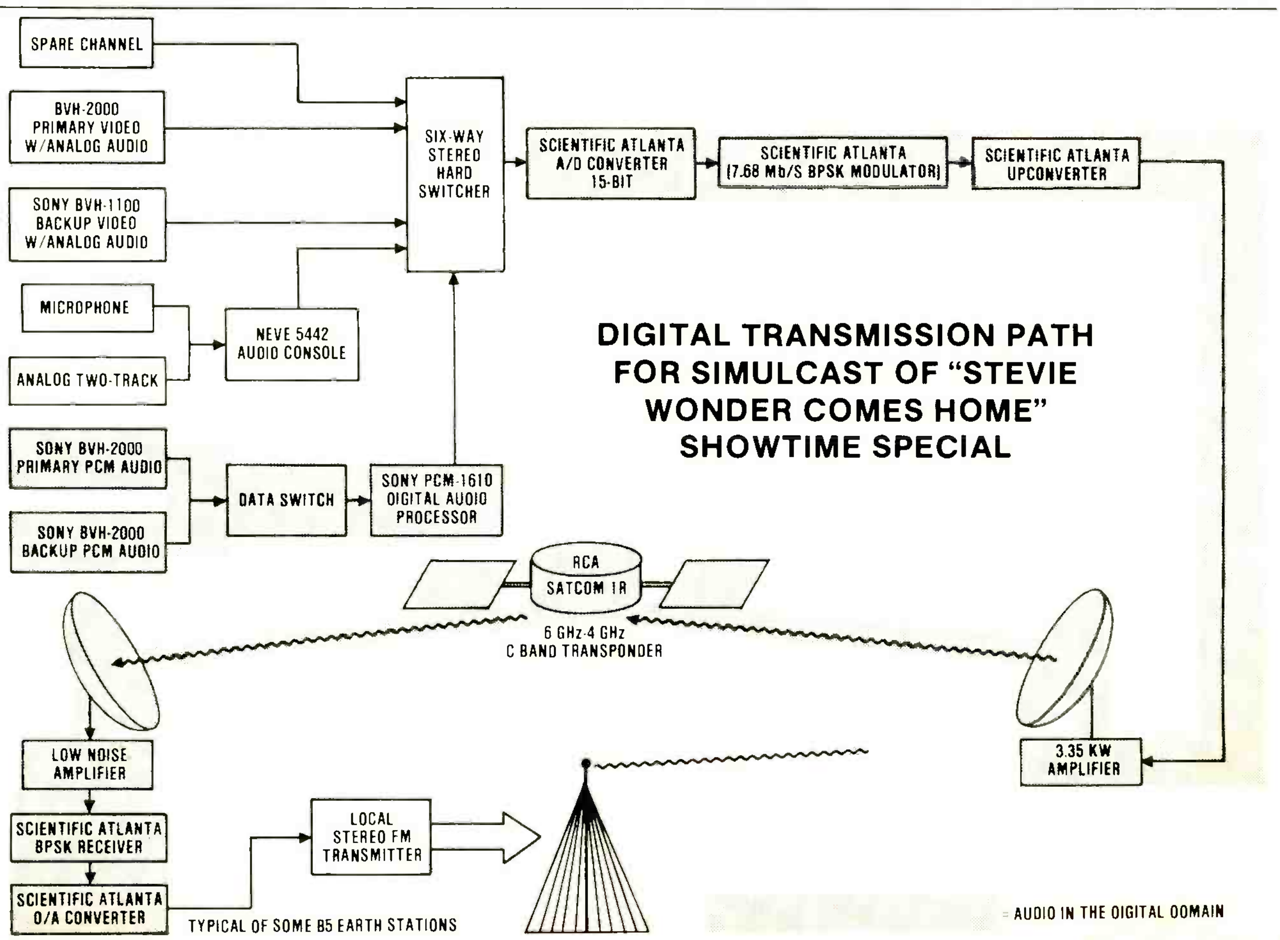
"After we finished the final day of recording, we copied all of the overlaps for safety's sake. So at that point, I had *two* complete clones, if you will, of every second of music."

Both of the 3M multitracks, as well as all of the videotape recorders, were fed the same SMPTE timecode and house sync from the New York-based Unitel Video truck. As a further precaution against missing audio program material, Neumann secured a Sony PCM F-1 digital audio processor, specially modified by Dave Smith of Editel [see April 1984 issue of *R-e/p*; page 110]. Neumann jam synced the F-1 to the house sync, and used it to record the stereo monitor mix in the Record Plant truck.

From the first, Neumann reports, the digital format had an impact on production procedures. It probably goes without saying that placement and selection of microphones, for example, were handled in much the same way as they would for an analog concert. However, the use of digital audio, the engineer says, makes everything more critical. "You *can't* hide behind tape hiss as you can with analog. You *have* to be all the more careful that everything is as clean as possible — no hums or buzzes. Some borderline things that would be acceptable on an analog remote were not acceptable here.

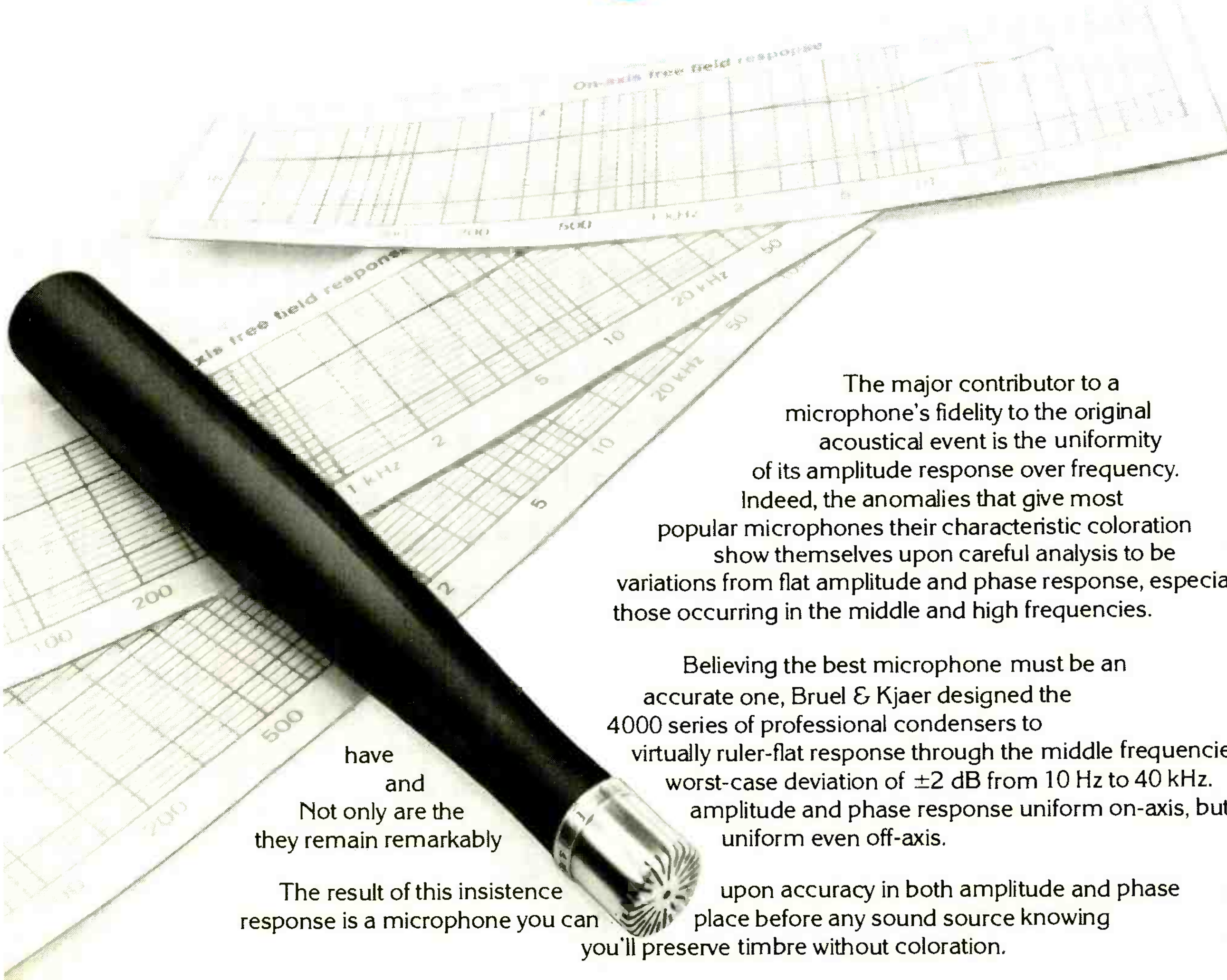
"On the positive side, I didn't have to do any compression or limiting at all thanks to the digital. A show like this is such a great thing to be able to do that way, because there's something about a live performance that you could never recreate in the studio; a certain excitement that you can only get in a concert situation. Not having to limit means that you can get it all on tape all the *more* live."

The desire to capture that sense of



A few words on microphone accuracy

from the people who specialize in it



The major contributor to a microphone's fidelity to the original acoustical event is the uniformity of its amplitude response over frequency. Indeed, the anomalies that give most popular microphones their characteristic coloration show themselves upon careful analysis to be variations from flat amplitude and phase response, especially those occurring in the middle and high frequencies.

Believing the best microphone must be an accurate one, Bruel & Kjaer designed the 4000 series of professional condensers to have virtually ruler-flat response through the middle frequencies, and worst-case deviation of ±2 dB from 10 Hz to 40 kHz. Not only are the amplitude and phase response uniform on-axis, but they remain remarkably uniform even off-axis.

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WORK IN PROGRESS

SONY PCM-F1 MODIFICATION TO ENABLE EDITING WITHOUT CONVERSION TO PCM-1610 FORMAT

A Co-processor-based System Currently in Prototype Form for Mute-free Insert and Assembly Editing of F-1 Digital Material

by David Smith, Editel/New York

The early months of last year saw a consumer product rear its impressive head in the United States and for the first time, almost anyone possessing a videocassette recorder could record audio digitally at home, a process that heretofore had been relegated only to research laboratories and a few wealthy recording studios. Little did the designers of the Sony PCM F-1 imagine the degree to which the unit would penetrate not so much the high-end home recording market, but the professional recording, motion-picture, and television production market.

Phase 1 of the F-1's "professionalization" has been well documented on the pages of *R-e/p*, covering such subjects as recording records, videodiscs, and movie soundtracks. Phase 2 of Editel's professionalization has been quietly progressing for some six to nine months; if and when it is completed we will have come full circle. Phase 2 is the heart of post production — editing — which is where our story begins.

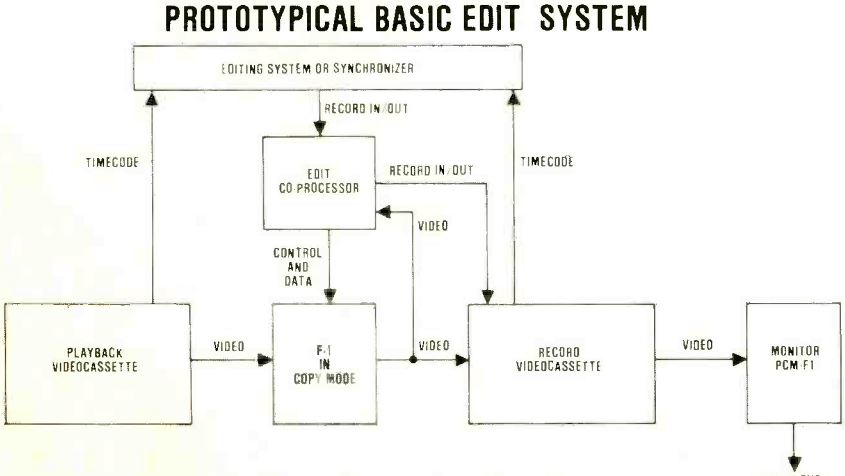
Almost from the beginning, people would hook up two videocassette machines and punch a piece of one recording into the middle of an existing one. The result, however, would almost always be the same: a pop or mute anytime an edit transition was performed. The reason for this can be found in basic digital recording theory, and appears to be based not only on economics but marketing strategy; the consumer digital format will not compete with the professional one.

Every frame of television contains two fields, and it is the packing of digital data into these fields that is causing our problem, since video editing joins two pieces of video by attaching the start of one field to the finish of another. When digital data is recorded on videotape, the various digital words intentionally are recorded out of order so as to minimize the effect of dropouts on the integrity of the recording. This process, known as interleaving, is reversed during playback as the signal is restored to its original order. Any burst (concentrated) error is randomized, thereby spreading the error out over a period of time, allowing the error correction to deal more effectively with the consequences of the dropout.

The basic unit of interleave is the Interleave Block; in a video edit, the end of one block must be joined to the beginning of another. All PCM F-1 editing today is performed by converting the F-1 format into the PCM-1610 format, for which there are exactly seven Interleave Blocks per field thereby satisfying our criterion for a successful edit. The F-1 format, however, has roughly 2.2 Interleave Blocks per field, and so there is almost no chance whatsoever that our video edit will be a success. (For example, if a field ends on *line* 29 of one Block and the field we are attaching to it starts on *line* 71 of another Interleave Block, the 52-*line* interleave error, 52 errors, will automatically mute the F-1 processor since the maximum number of interleave errors it can handle is two. By the way, a video field contains 262 lines of video — half the NTSC 525-line rate. PCM-1610 Interleave Blocks are 35 horizontal lines each, while the F-1 Blocks are 113 lines each.)

Rather than attempting to deal with the problem from a video standpoint, the computer-science people are treating the F-1 as a source of data, which is extracted

PROTOTYPICAL BASIC EDIT SYSTEM



DIGITAL RECORDING AND POST PRODUCTION

"event" at the show, combined with an appreciation of the good acoustics at the Masonic Temple — a traditional proscenium arch theatre with draperies and upholstered seats — led Neumann to devote great care to audience miking. He used a combination of Schoeps mikes set in an X-Y configuration, and an Audio+Design/Calrec Soundfield system, which utilizes four cardioid capsules arrayed on the faces of a tetrahedron. A separate control unit/matrixing system enables the recorded B-Format signals representing a three-dimensional Ambisonic soundfield image to be manipulated in post production.

"You can actually steer the image after the fact," explains Neumann. "You can turn it around, move it left and right, up and down, or make it wide or narrow. Of course, the system works with analog machines, but it works so much *better* with digital.

"Which is one aspect of digital that people usually don't dwell on. Everybody thinks of such features as noise elimination when they think of digital; but one thing they often fail to mention is that there are *no* azimuth errors or tape-path problems. It makes an immense difference with the Calrec system.

"When you're de-matrixing [the B-Format signals], the image is *incredibly* stable. You can pinpoint people in the audience, and they stay there. There's no ambiguity right up to the top of the frequency spectrum. It's just *rock* solid."

Audio Sweetening and Remix

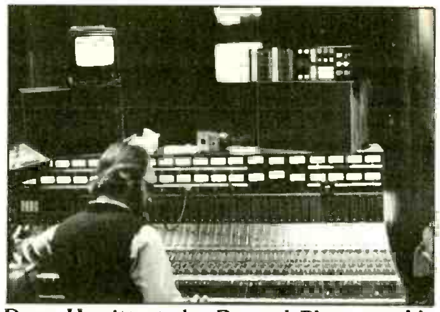
Audio post production for the show began at Wonderland, Stevie Wonder's own Hollywood facility, and then moved to nearby Westlake Audio because of scheduling commitments at Wonderland. The audio sweetening and remix did not begin until a final picture edit had been prepared at Complete Post in Los Angeles. Along with the finished picture, Complete Post provided a four-page printout of

Inside the Record Plant/N.Y. mobile: two Sony PCM-F1 processors for digital backup, and an Otari MX-5050 four-track to replay various music and effects tracks.

MARGIE STROBEL

all the video edits, generated by the facility's computerbased editing system. These video edit notations then became the basis for conforming and assembling the audio tracks.

Among the items of information included in the edit decision list were two sets of timecode numbers. A master code provided consecutive addresses from the beginning to the end of the complete show, while a production code furnished addresses for the individual segments that were edited together to create the complete show. Using combinations of these notations, assembly of the final audio track began begun. As it turned out, the majority of tracks were taken from the final night's concert, which everyone involved with the project



Dave Hewitt at the Record Plant truck's modified API console, with additional MDM-4 Near Field monitors and remoted 3M DMS peak-reading level displays .

felt was the best performance by far. Material from previous evenings was used in lieu of overdubs to "fix" mistakes on the part of the performers, by transfering various takes via code offsets.

It was necessary to design some custom synchronization equipment in order to make 3M's digital editing system compatible with the SMPTE timecode-based videotape master, and capable of performing the audio edits. "The 3M works fine for what it was designed to do," explains Neumann. "It was *not* designed to do post production for video or anything like that. Therefore you don't access your edit points on it via SMPTE timecode. Instead, it has its own 3M edit code, which has nothing to do with SMPTE. It is displayed in minutes and seconds; there are no frames, as in video systems."

The solution to the problem was a custom-designed external clock for the machines, devised in large part by Frank Dickenson. The custom clock worked in conjunction with an Audio Kinetics Q.Lock synchronizer, which switched the machines back and forth from their internal clocks to the custom external clock common to all of the machines. In this manner, the 3M multitrack and four-track digital mastering machine were fully synchronized to each other, and to the timecode-bearing master videotape, played back on a Sony BVU-200 U-Matic.

Video-follow-Audio Editing

Although it is standard television practice to complete picture editing before assembling and conforming the associated audio tracks, according to Neumann this procedure created a few editing problems on this concert program. "There were several edits that worked fine on video, but which were *really* problems for the audio," he recalls. "There were several occasions where we had to spread out an audio edit. Some of the tracks would happen right at the edit point, but others had to be punched in a little before or after. Say there was a drum part that didn't work right on the edit; maybe there was a little 'flam' or something. Or maybe there was a little pickup or breath on a vocal, and it had to come in before the edit point.

"What we would do was get our edit point happening, and then do one pass for those tracks which fell before the edit point, a second pass for those

A custom-designed subconsole linked to the Record Plant's main API board handled additional stage and audience miking during the Stevie Wonder videotaped concert.

DIRECT EDITING OF PCM-F1 TAPES — continued . . .

from the processor before it is interleaved. This serial data-stream is recorded either in RAM or on floppy disc, where it may be manipulated, level corrected, and edited with impunity, after which it is returned to the processor for interleaving, generation of error words, and recording. This approach works extremely well if you have a microcomputer, and the associated software to accomplish the task. The Editel division of Columbia Pictures, however, possesses a wealth of video recording and editing systems, as well as several clients recording their audio-for-video on an F-1, leading us to try to solve the riddle of F-1 video editing without resorting to the development of another specialized editing system.

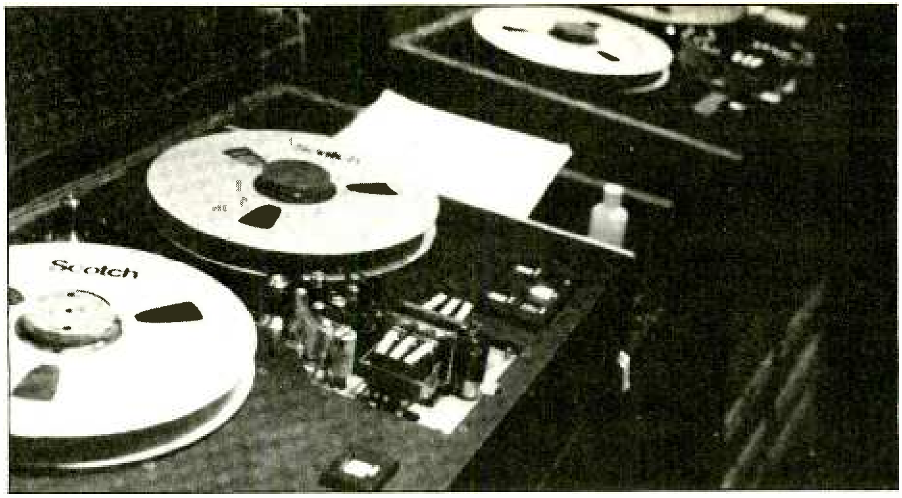
Our approach has centered around the characteristics at the heart of the F-1's playback side: the TM-4505 playback processor. This large-scale integrated circuit, together with 16K of RAM and several associated circuits, de-interleaves the data coming off tape, corrects errors caused by dropouts or timing jitter, and outputs the data to the D-to-A converter for conversion into the analog domain.

What we at Editel have developed is a co-processor that allows us to video edit out of the middle of one Interleave Block and into another almost anywhere, without any audible noises and without the use of a sophisticated microcomputer and its requisite software.

At present, this processor is in prototype form, but shows great promise as far as allowing us to edit F-1 audio on any video editing system synchronously with picture, or just on two videotape machines for simple music editing. To date, the system has been used to edit musical sequences for several episodes of *The New Show*, and to edit Sony's Audio-Visual presentation at the 1984 NAB Convention, *Fast Forward Into The Future*.

As it now functions, our co-processor controls an F-1 in the copy mode, this F-1 receiving its input from the player and outputting its signal to the recorder; a second F-1 decodes the output of the recorder for monitoring purposes. When the operator or editing system places the edit machine in record mode, the co-processor generates an edit or transition signal that is recorded on the first 10 lines of the first field to be recorded, using the F-1 in copy mode to place the edit signal into the video data stream. The same process is employed as we arrive at the out point, the last 10 lines of the last field to be recorded containing the transition data. Needless to say, the co-processor has its hands full counting video lines, determining fields, generating data, and watching for the record in and out points. By altering the transition data and the number of transition words recorded, we have been able to alter the characteristics of the edit, but this is still a grey area.

At the time of writing, the only thorn in our side is editing extremely low-level material. This is to be expected, however, since we are trying to bend an immutable set of laws that have been etched in silicon, namely the operating parameters of a large-scale IC. The final stumbling block is not insurmountable, and we feel that our attempt is worth the effort. The execution of Phase 2 of professionalizing our F-1 justifies the effort, especially if it will eventually be adaptable to anything from a VHS portable to a one-inch Type-C helical videotape transport. □□□



Two M81 digital 32-tracks were rented for the concert shoot from Digital by Dickenson.

which fell right on it, and another pass for those which fell after. A lot of it involved quite a bit of trial and error. By no means was this a matter of simply plugging in the edit list and sitting back — each edit was agonized over!"

Although there are ample console automation facilities at Westlake Studios, where the show was mixed, Neumann preferred to remix the show manually. He mixed in 20-minute segments (the 3M digital reel holds a maximum load of 30 minutes), working in sequence from the beginning of the show to the end.

Audio was mixed to picture, something in which Neumann is a firm believer. "It was very important to be able to have the picture there while we were mixing. Even though I know the show very well, on the occasions when I'd try to mix without the picture — for whatever reason — it would never work right when the audio was put back together with the picture."

Once the mixing was completed, the show was brought to Burbank's Compact Video for sweetening and layback to the final master formats.

"Given that this was a TV show, it was destined to have sweetening. But I must confess that, at the outset, I was philosophically opposed to it," Neumann admits. "I was glad we did it in the end though; it helped the show. They had classically miked digital recordings of the actual audience and the sweetening was done on digital. So it wasn't like they were flying in laugh tracks from *The Johnny Carson Show* on a cart machine!"

Digital sweetening was accomplished by transferring the mixed stereo tracks from the digital four-track back onto the digital multitrack. "We took our digital multitrack to Compact and had them do their sweetening on open tracks of the multitrack," Neumann explains. "Our mix and their sweetening were combined. The final composite mix was to a Sony PCM-1610 [and one-inch videotape] with timecode on what would have been an analog audio track. It had been decided that the Sony system would be what we would use ultimately for the simulcast. The fact that it is a video-based system seemed to lend itself to our purposes. Also, we had a one-hour show; unfortunately, the 3M [four-track] can't hold a one-hour load."

Layback, also handled at Compact, involved several different tape formats. Along with the digital audio two-track master for FM simulcast, stereo and mono analog tracks had to be provided for Showtime. The cable company's format called for stereo audio on tracks one and two of a one-inch videotape, mono audio on track three, and timecode in the vertical interval.

It was a track configuration that gave Neumann some cause for worry. "The control track is immediately adjacent to the third audio track on the one-inch format," he explains, "and fringing at low frequencies on the control track — caused by kick drum, bass, etc. — can actually throw the video sync off. I was also a little concerned about the peak content going to the analog tape. It looked like we were going to have to compress and limit the signal, filter it, and do all kinds of undesirable processing. But we first tried the transfer without all of that; we just used Dolby A-type noise reduction on the stereo tracks. We got lucky, and it worked without the processing."

NOISE REDUCTION APPLICATIONS IN DIGITAL AUDIO

Although the noise-reduction industry grew up around analog tape technology—and without noise reduction the development of multitrack recording would have been severely hindered — it seems to be making a smooth transition into the age of digital audio. The two leading manufacturers of noise reduction — Dolby and dbx — have developed A-to-D and D-to-A converters that incorporate elements of each company's respective analog noise-reduction systems.

Each of the two systems is aimed at a different application; Dolby DP-80 Digital Audio System is designed to improve the quality of satellite transmission, while the dbx Model 700 is primarily targeted for recording applications. Both systems, however, employ variants of Delta Modulation — as opposed to PCM (Pulse Code Modulation) that thus far has dominated the digital field — as the digitizing process.

"Delta Modulation is inherently less complex to use than PCM," comments Dolby director of communications, George Sell. "We have developed a means of using it that eliminates some of the problems traditionally associated with Delta Modulation, enabling us to take advantage of its greater simplicity and ease of use.

"Basically this advantage stems from the fact that, in Delta Modulation, all the bits in the bit stream are of equal value. In PCM, they have different values, which makes the circuitry more complex, and therefore more expensive. Because of the redundancy in the [PCM] circuitry, you have a wider bandwidth to transmit; among other things, you're sending extra bit streams that are strictly transmission information, rather than program information.

"All of this makes PCM more expensive; and as cost-effectiveness was one of our prime concerns in developing the system, we decided to explore Delta Modulation instead."

Prior to digitization via Delta Modulation, the analog signal is treated with a type of noise reduction Sell describes as "a further evolution of the Dolby B-and C-Type noise reduction systems developed for analog audio recording."

As a further cost-effective measure, Dolby has engineered most of the system's complexity into the encoding equipment, enabling them to market inexpensive decoding equipment. Among the firms licensed to use the system are General Instruments, Oak Communications, Racal-Oak (a UK-based operator), and Wegener Communications (MTV, Showtime, and The Movie Channel).

In the dbx Model 700 digital processor, Delta Modulation is utilized in conjunction with a dbx spectral compansion noise-reduction system similar to the one used in the dbx/Zenith Stereo Television broadcast system. Spectral compansion is a relatively new noise-reduction concept that dbx has added to its traditional amplitude compansion with fixed pre-emphasis noise-reduction design.

"The reason for the spectral compressor is to vary the pre-emphasis in the audio channel to suit the needs of the program material," explains dbx vice-president/engineering Les Tyler. "For example, if the program material consists primarily of high frequencies, these high frequencies have a chance of overloading the channel if the amplitude compressor pulls the level up too high. If the high frequencies do not overload the channel, they are likely to effectively mask the noise of the channel [the goal of all noise reduction], in which case you don't want or need to have a lot of pre-emphasis.

"When a *very* strong high-frequency signal comes along, the spectral compressor will actually reduce the amount of high-frequency content in the channel. On the other hand, if there isn't much high-frequency content in the program material, then the spectral compressor will push up the level of whatever high frequencies are present in order to mask noise. When you play back or receive the program material, you can turn those high frequencies down in decoding and the noise will be reduced."

Tyler foresees widespread possibilities for spectral compansion noise-reduction beyond its present applications in either Stereo Television broadcast or digital encoding/decoding. "We have been discussing use of the system with some cable companies, and some of the people involved in direct satellite transmission. This type of system in such applications is something that makes a lot of sense to all of us. It is an unusually powerful noise-reduction system, and shouldn't be regarded as limited to any one area." □□□

Digital Simulcast Transmission

Simulcast transmission of the completed show was handled by Westwood One, Showtime's syndicator. "Westwood One had their choice of several satellites that they could have used," Neumann reports. "The RCA SatCom 1R is the one we ended up going with. It is a Scientific Atlanta digital system; a very fine system by the way. Not only were we feeding them ditigal audio, but the transmission up the satellite and down to all the affiliates around the country was digital as well."

As part of an overall effort to keep the simulcast as clean as possible, it was decided to bring the digital processor and video playback machines to RCA's uplink facility in Vernon Valley, New Jersey, thus avoiding microwave links between the uplink and any studio site. Neumann flew to New Jersey to supervise the preparations and transmission. Since this was the first time that a digital audio simulcast of this nature had been attempted, the engineer naturally felt he should be on hand to attend to any last-minute problems that might arise.

In at least one respect, this precaution paid off, as Neumann explains: "As it turns out, after all the trouble we went through on the audio for this,

Supervising engineer Lon Neumann (right) and Sony's Gus Skinas at the RCA satellite uplink facility in New Jersey for last June's digital-audio FM/Showtime simulcast.


Four Sony BVU-2000 VTRs provided main and backup video replay plus PCM digital audio playback at the satellite uplink site.

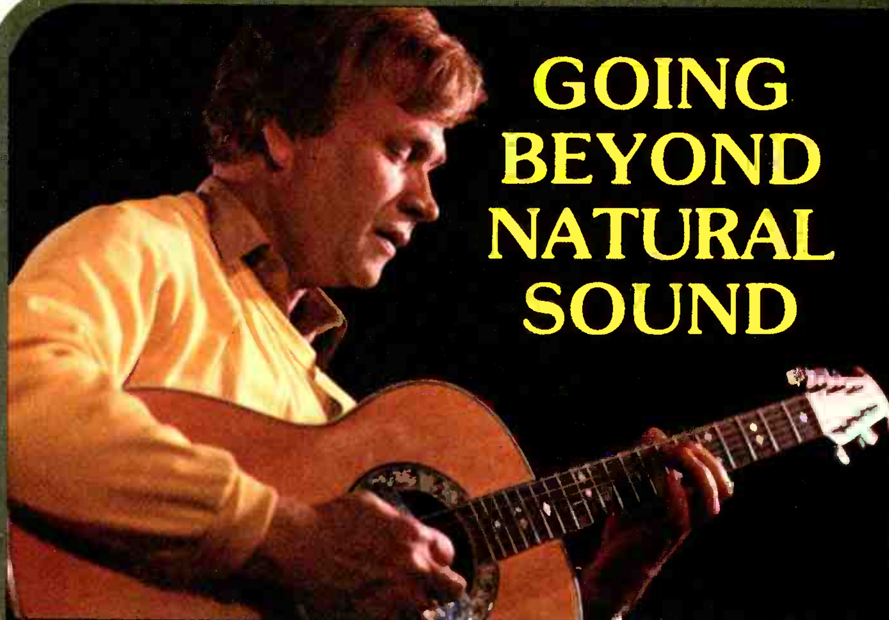
at the end of the chain they wanted to take the whole thing through an inexpensive little mixer, just so they could do their voice-overs at the beginning and end of the show.

"What I did was talk to the program director and convince him to rewrite his copy so that the announcements didn't have to go over the show; they could come before and after. I then just built a switch box that would enable us to do a hard switch between the announcer and the show."

Sony furnished four BVH-2000 one-inch, C-Format videotape machines to be used for playback at the uplink. Two transports were assigned for the main and backup video replay; tapes on these machines also bore analog audio tracks. The remaining two video machines provided main and backup digital audio playback, which was processed through a PCM-1610 and uplinked in digital form to the satellite. Neumann built another custom switch box to enable him to switch from the output of the main audio machine to the output of the backup machine, in the event of a failure. As it turns out, however, this precaution wasn't necessary.

In all, Neumann reports that he was quite pleased with the show and its transmission. "If nothing else," he says, "I hope this project will at least demonstrate that digital audio-for-video works. No matter what kind of release format you have, it's much better to keep a project digital *all* the way. You can hear the results in the sound, and the editing freedom digital affords is incredible. There were so many points along the way where it was right to make a copy. But with digital, you don't make copies, you make clones; nothing is lost. There is also the archival advantage offered by digital — everything stores well.

"I ran into a lot of skeptics in the course of doing this project, but I think the show will convince them that digital works. People are talking about high quality audio for video quite a bit now; *this* is it." ■■■



Photography by Kathy Cotter

GOING BEYOND NATURAL SOUND

RECORDING AND PRODUCTION TECHNIQUES FOR A UNIQUE PROJECT DESIGNED TO DUPLICATE NATURAL SOUNDS OF WELL-KNOW GUITARISTS

by Jimmy Stewart

Using some of Los Angeles' top-flight studio jazz musicians, the author recently produced an album entitled *The Evolution of Jazz Guitar* to serve as a companion piece to his latest book, *Jazz Guitar*. Recorded at Monterey Recording Studios in Glendale, and mixed at Sound City in Van Nuys, California, *The Evolution of Jazz Guitar* presents 14 self-penned tunes written in the style of leading jazz guitarists whose careers spanned almost half a century. Working with recording engineer Chris Minto, the author attempted to recreate the sounds of the Twenties through the Sixties, but with an Eighties twist using modern mixing techniques. The album can be considered somewhat unique in that it merges the time-honored jazz recording tradition of pure, natural room sound with Eighties recording technology, to create a new blend that is of its period, as well as more "natural than natural." This article describes some of the recording and production techniques used on the album project.

During preparation of this album project I attempted to combine the best of multitrack recording techniques with the best of the Jazz, live-sound concept. I wanted to achieve a stereo image of the various composite musical elements that would blend in the final mix — piano, drums, bass, horns and guitars — as well as capturing a true representation of the room in which they were recorded. Furthermore, the recording enviroment had to be *complementary* to the natural sound of the acoustic instruments when it was recreated in the final mix.

Our goal was to recreate, on a single album, the styles and sounds of all the major jazz guitarists that have emerged during the last 50 years. But recording techniques have changed dramatically over the last five decades. In the old days of recording, one mike was used in the studio and there were no re-takes. The sound of the artist was the best that could be captured with a limited frequency-range mike. Since I was going for an Eighties' equivalent of that original sound character and performance, the first decision was to find a good-sounding room and interface it with modern-day technology.

Studio Selection

My recording engineer on the project, Chris Minto, and I chose Monterey Studios, Glendale, California, for its flexibility, which would enable us to recreate the different environments and sound textures. The monitor system was Chris' first consideration, and the recording environment mine. Our check list ran as follows:

1. Sound of the studio — an open, pleasant-sounding room.
2. Monitor and foldback systems — an accurate system upon which to base our creative decisions.
3. Metering — to check our transient material, and maintain a wide dynamic range on tape.
4. Microphone selection — clear and clean-sounding mikes.
5. Hardware and application — a clean recording chain.

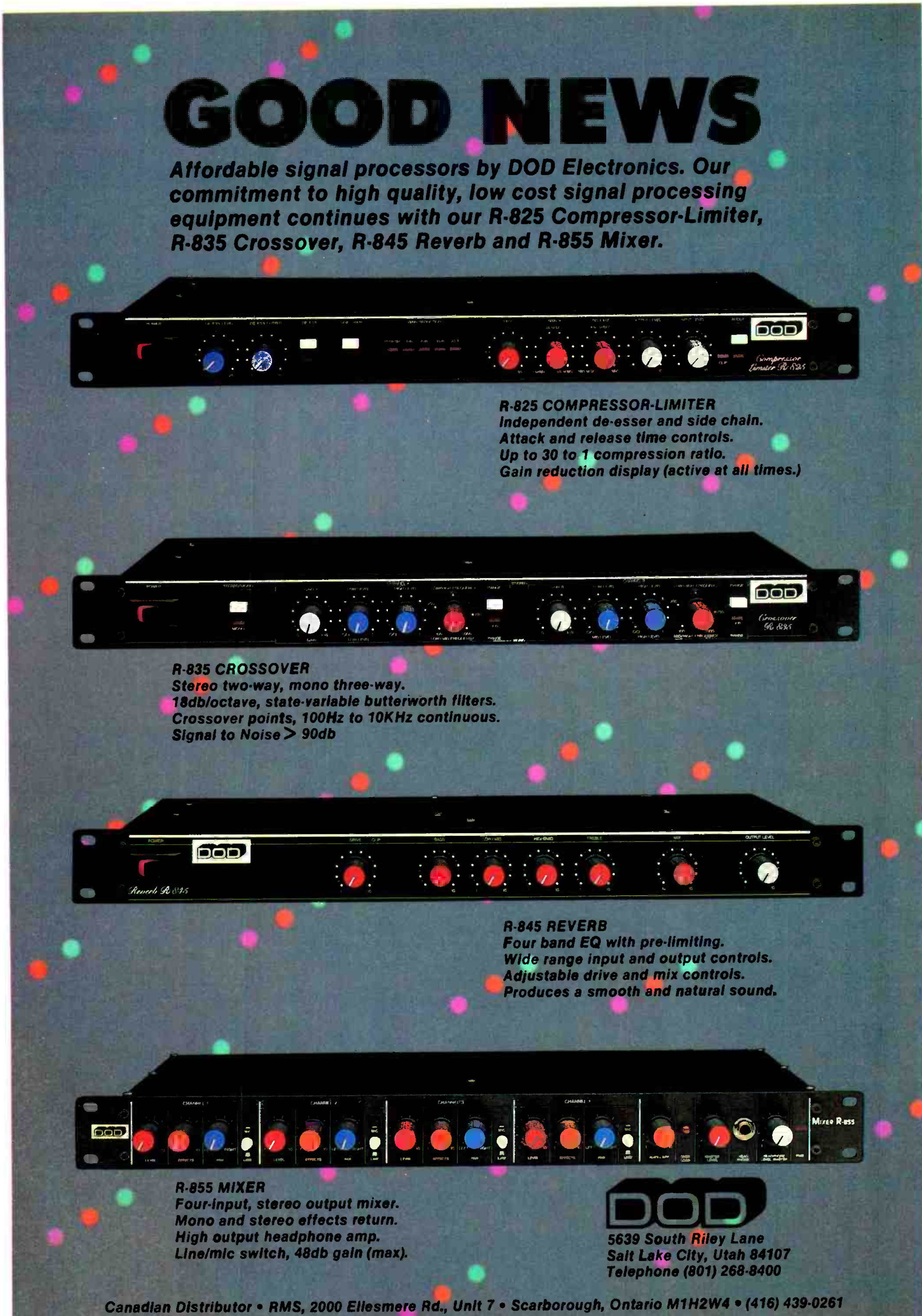
To test out the room's ambience, I recorded my voice in the studio on a portable Sony cassette recorder (using its built-in condensor mike) and played it at home, listening to the room reverberation quality, and texture of the sound. Chris made a tape from an album he recently recorded, for use as a monitor test in the control room. I also played his tape in my living room and personal-use studio, to set up an acoustic benchmark for our listening tests.

We took the tape to different studios were considering buying time from and, as we listened, Chris encouraged me to tell him what nuances of sound I was hearing. He would point out in technical terms the pros and cons of each monitor system, and what we could achieve from our sounds.

In the studio area, I was looking for a flexible environment; one that was capable of providing different textures to different instruments. When we finally started to work at Monterey Studios, I ended up using every area but the restrooms. Some parts of the studio I liked, and some I didn't. Within the recording budget I had allowed myself the luxury to explore what sound we could achieve by experimentation. Such flexibility is very important when you work in a studio for the first time; I knew the sound I wanted, and by experimenting, I eventually got it.

Beyond Natural Sounds

I wanted the instruments to sound natural in the room but, at the same time, to get a little *beyond* a natural sound; I wanted to put the listener's



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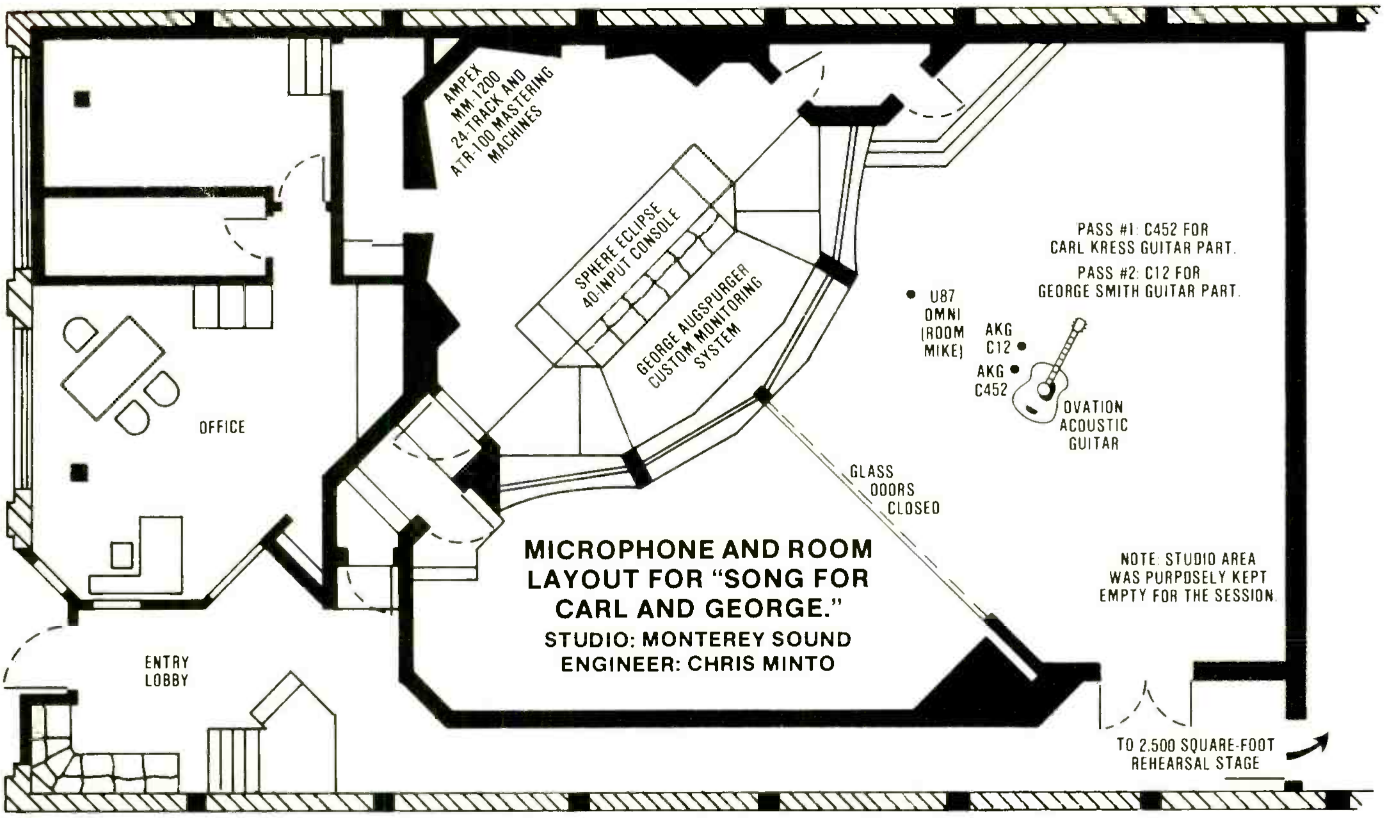


MICROPHONE AND ROOM LAYOUT FOR "SONG FOR CARL AND GEORGE."
STUDIO: MONTEREY SOUND
ENGINEER: CHRIS MINTO

GOING BEYOND NATURAL SOUND

ears inside the instrument to provide a different point of view. Coming from the production side, I called on my previous experience as a musical director standing in front of an orchestra in a concert hall. When I used to conduct for different artists, part of my preparation plan before he or she arrived for the run through was to check the music balance as the audience would hear it. I'd start the orchestra on a tune that didn't require a great deal of direction, and then walk around the hall to get a feel for the ambient sound. If we used heavy sound reinforcement this was an even more critical check. As a music director, my responsibility was music, sound an lights; in reality I was the producer or "framer" of the show. If those elements were not just right, my clients let me know in the dressing room.

For the album project we were attempting to create an overall sound, as if we were going to watch a performance at Carnegie Hall. We wanted to sit right in the third row center and feel that the guitarists were sitting right there with us; and, at the same time, have the depth and integrity of a wonderful sounding concert hall.

Sitting as a listener in the third row of a concert hall, you hear the direct delayed sound, then a blend of reflected sound from the walls and materials found in the hall. From this perspective the listener hears a combination of what is heard from the conductor's point of view, along with the hall ambience. It's like being bathed in the sound.

A similar concept was also used when setting up the players' foldback system. Even though we didn't print the delays and echoes on the track, we added these discrete mixes to the foldback headphones. In fact, in terms of relative volume levels between instruments, the foldback mix was identical to what we were monitoring in the control room.

In terms of the tracking dates, having a glass partition across the more-dead sounding side of Monterey's room was very important for us, because we could use it to liven up the sound of certain instruments, and to provide separation. We ended up putting the piano in this separate room, so we wouldn't have unwanted drum leakage to contend with. In the old days when engineers were cutting jazz albums, they simply put everybody in the same room and worked with the leakage. Remember that if the musicians are put farther apart in the studio, the leakage — through reflections of various surfaces, as well as path-length differences — has time to get out of phase, and tends to detract from the direct sound. Whereas if you put people closer together, the leakage tends to be additive.

Modern techniques seem to employ people being placed in separate areas to eliminate the leakage all together, and then to specialize on each individual sound, which is what we did with the piano. To keep it away from the drums, acoustic guitar and backline amplifier, we decided to place the piano in a separate room to give us a cleaner recording.

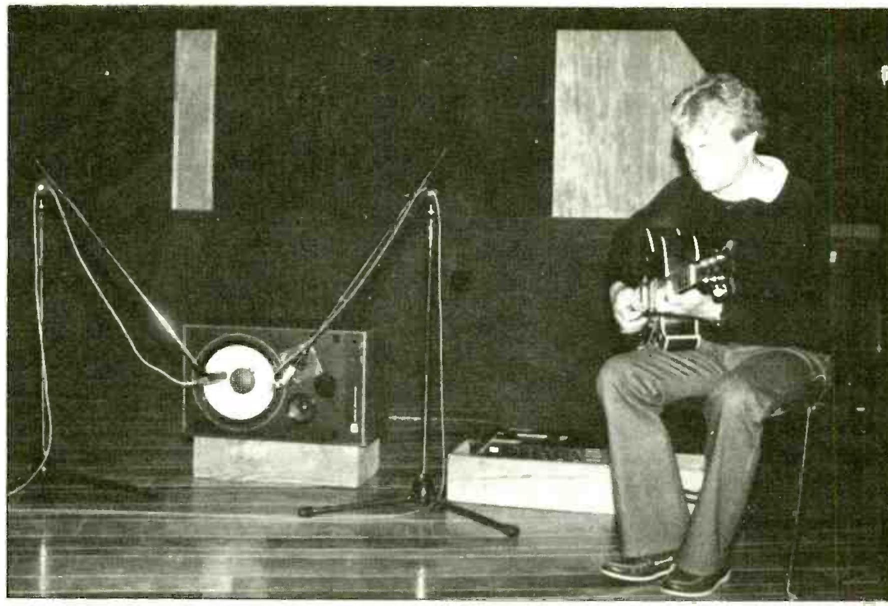
Live but Controlled Acoustics

Basically what we were looking for as a recording environment was a very low ambient sounding room. We wanted a room that sounded bright: a space that had a certain degree of reverberation, but then a quick, even reverb decay without standing waves. In that way we could put the guitar in the center of the room with a close mike, and then use a distant mike if we wanted to capture some of this reverberation within the room.

We attempted to portray each instrument larger than life, which essentially comes from the process of close-miking. However, that's only part of the story, since nobody listens to an instrument with their ear about an inch away from it; instead, you normally listen several feet away. So, when you close-mike an instrument or backline amp you get an artificial intimacy, a "hugeness," that is very pleasant and which records well. Sometimes, however, it sounds a little hyped, and so you need to incorporate such close miking with distant-miking techniques to add enhanced perspective. If you don't make use of the natural room sound, you can simulate it with echoes, digital delays, and effects of that nature.

Close-miking did ensure the integrity of the final product as we went through the different multitrack production stages. If the instrument is small-sounding to begin with, as it is transferred from multitrack to two-track to disk, there will be various losses of low-level ambience information, which loss can be overcome by emphasizing the direct sound from close miking.

Sometimes we blended in the room sound while recording certain instruments, such as the horns. Although the guitar tracks had room sound blended in with the direct sound, on top of that we would add straight reverb. Then, to obtain an open, "spacious" feel, we would use a delayed reverb, so there would be an initial slap, which would mimic the bounce or slap echo that is heard in a concert hall as the sound bounces off the back wall. Then the reflected sound becomes reverb as it bounces around the room, and you hear that reverb pattern decaying. If you only hear that reverb, without the initial slap, you don't have that sense of depth you would feel in a big concert hall. So you have to have *both*: the slap or delay, *and* the reverb.



Studio microphone technique for "Johnny's Mellow Mood," with a Shure SM57 on the JBL 4311 cabinet's bass speaker, and an AKG C452 covering the MF/HF driver.

When the tracking session first started, the room environment was considered integral to the recording process, because we were using distant mikes. In the late Sixties, the introduction of multitrack recording meant we needed to go for increased acoustic separation between instruments, and close miking became the accepted technique. At that point studio designers started to use more and more sound absorbent materials to deaden the sound, and recording environments became very sterile.

But now we're getting back to using the natural sound of a room, and utilizing a combination of close-miking and room-mike techniques. Many producers, myself included, see an increasing trend towards mixing natural sounds with synthetic sound, combining close miking with room ambience, and the cross relationship between the two. You can draw from two different areas — record one set of instruments one way, and another in a completely different way — and incorporate those textures to provide something that was impossible to achieve a few years ago.

Individual Sound Textures

We were obviously trying to create different "spaces" for each instrument; to create a different sound for each piece, and possibly give each one a different texture by placing it away from other instruments, so that it would stand out by itself. Oftentimes it seemed that this approach was simply the most honest way to record that particular instrument. When that sound becomes enhanced, the additional reverb is generally a touch more overpowering than the natural room sound, but the latter just gives it

ESTABLISHING A CREATIVE WORKING ENVIRONMENT FOR STUDIO MUSICIANS

To set up the musician's foldback system, I scheduled the first recording session to track the most extreme compositions included on the album: a high-energy, classical guitar piece over-dubbed with three synthesizers; and a softer tune for acoustic piano and jazz electric guitar. Working in this way I tested the control-room equipment's noise floor with the piano and electric guitar. With the classical guitar I started listening for locations in the studio that had the necessary acoustic support. My engineer Chris Minto and I also worked out imaging for the foldback — the same one that was used later in the mix. In the next session we concentrated on the drums and bass sounds.

Every aspect of the production was documented, so that we could pyramid into the larger ensemble groupings. With detailed documentation, we could always reconstruct the previous session, right down to the foldback mix.

To begin our preparation work, the crew and I would arrive 2½ hours before the players. Chris Minto, assistant engineer Phil Brown, and I would check out every phase of the recording chain. By the time the players arrived, the cue mix from the previous session would be playing in their headphones. Every headphone cue was played at the same volume level, so that while they were setting up for the session, each player could get into the vibe.

The foldback and playback mix are the most important elements for a musician while capturing his or her performance. The sound engineer and I take the time to discuss our approach so that, as they are playing, the musicians hear a very close approximation to what will be recreated in the final mix. Imaging and effects for the acoustical environment will be used in the foldback and playback mix, even though they may not be printed to tape. This technique gives the player a feel for what we are trying to achieve in final mix. Musicians respond to sound, telling them what you are *going* to do with their sound isn't enough. If you want a good performance, take the time to let them hear where you are coming from.

Since I was serving as primary musician *and* producer, for the foldback mix Chris provided us players with a perspective of being "under the microscope," as opposed to a less detailed cue system. Some of the players were expecting a jazz-type of sound concept in the headphones, but it wasn't. They got to hear clarity *and* imaging; a playback and foldback system that had a balance between the two. As a result, even players performing in a separate room — the horn section, for example, was recorded on the second floor, while the rhythm section was on the first floor —could hear exceptionally well. With current techniques it is possible to achieve that kind of definition and clarity in the earphones *and* on tape, even though you're cutting all the instruments at the same time. □□□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GOING BEYOND NATURAL SOUND

a different texture. The end result is a different sound from that obtained by simply using a close-miked sound with added reverb. It's not bad or good; it's just another *color* — another taste for the listener.

However, if we had, for example, natural room reverberation on the miked sound of a backline amplifier, and had then placed too much additional reverb on that track, we would have been in trouble, because the combined result ends up sounding mushy and confused. While cutting guitar tracks we added a touch of artificial reverb — just to give the sound some life — because close-miking can give a rather stark sound when you don't incorporate much of the room sound. So, you put in a little artificial reverb to synthesize a little more of the natural room sound. But it's equally important to use artificial reverb that *complements* the natural sound of the room, and not make it counteract and conflict with the sound you are after on the track.

In the everyday use of recording techniques there are certain ideas and methods that you use because you know they work. But you can employ a whole bag of other tricks. As we were going through the process of trying to frame each instrument on the album, the players and I would talk about a certain sound, or a certain concept we wanted to go for. Out of that process will spring an idea. Sometimes it wasn't a technique we've ever used before but, by incorporating bits and pieces of other ideas, you develop something new.

The track, *Johnny's Mellow Mood*, was written for guitarist Johnny Smith, a studio jazz player from the Fifties. Striving for a better, unique guitar sound, Smith developed the Johnny Smith Guitar made by Gibson. This model featured a single pickup mounted close to the fingerboard to produce a rich, mellow tone bringing out the midrange frequencies.

As an example, for the *Johnny's Mellow Mood* recording we plugged the guitar directly into one of the studio's amplifiers — a 100-watt BGW amp that sometimes is utilized for the cue system, but wasn't in use at the time — and then into a JBL Model 4311 speaker. The BGW/JBL combination produced a *very* clean sound. We mixed the sound of a condensor aimed at the high-frequency driver, and a dynamic at the low-end speaker. The result didn't sound like a direct-inject guitar — which has a whole other feel to it — but rather the clean texture of the amp and the speaker combination.

I think that the recording of acoustic instruments is a great place to start on a project because, in some cases, electric instruments are trying to synthesize what acoustic instruments do naturally. So, if you have a feel for the way acoustic instruments sound — acoustic guitar, classical instruments, violins, violas, cellos, brass, and those kinds of instruments — you have a perspective on what electronic instruments are trying to synthesize.

Microphone Selection

Another interesting aspect of the project was that we used different microphones and miking techniques every time we recorded the guitar, each one being unique to the style we were trying to capture. Although we might be recording the same guitar, employing a different mike technique provided a different texture and feel. This approach proved particularly useful when we were doubling the guitars, but still wanted to retain a different identity for each instrument. There might be three parts, for example, each played by the same guitar. If you used the same technique of miking for each one, it wouldn't be as easy to define the subtle differences, or for the listener to image each different part. By employing different texture for each part, it is easy for the listener to feel that he or she is hearing three distinctly different guitars.

Take as an example the *Django Reinhardt* passage we recorded for the album, which consists of two rhythm parts that were panned to the left and right sides of the stereo soundfield, and were recorded on two passes using a Shure SM57. Then for the lead sound, we employed a little bit of the Neumann U87 room mike, blending it in with the SM57 to add a little bit of a different texture and space. Early studios were small and acoustically dead, and during the Thirties and Forties live radio sounded better than the sound of a record.

Reinhardt was a Belgian guitarist who left a legacy of recordings and transcriptions from radio broadcasts during the 1930s. Two members of his family also played guitar in the original quintet of the Hot Club of France. Chris and I wanted to portray the difference in the three guitar sounds that could not be heard on radio broadcasts or recordings made during this period.

The inherent quality of a microphone determines the choice of mike for a particular instrument. As an example, condensor microphones work very well on instruments that need ultra-clarity, or that produce sharp transients. Dynamics work well if you need to have a certain degree of sonic clarity, but also want a "warmer," almost dull sound. And you'll turn to a ribbon mike if, for example, you want to capture the "splatter" from a trumpet. We used a Coles 4038 on cornet — which worked beautifully — whereas a smoother, rounder-sounding Electro-Voice RE20 dynamic worked beautifully on tenor saxophone and a Neumann U47FET on baritone sax.

In terms of acoustic guitar, some mikes enhance the sound better on certain models of guitars than others — it really is a matter of putting up a microphone and listening to the instrument in the room. Then you can have your engineer go back into the control room and tape a few bars, and have him play it back for you to see if it

GOING BEYOND NATURAL SOUND

sounds the same as it did in the studio, or that it sounds how you want it to sound. And, if it doesn't sound quite right, change the microphone until you find something you like. When you get close to the sound you're after, it usually requires a *touch* of equalization. But you shouldn't try and save it in the control room by doing massive amounts of gymnastics on the console's EQ section.

For the room miking we used omnidirectional microphones, because that's exactly what we were after: the total sound of the room. We used cardiod patterns for the close mikes to prevent excessive leakage from other instruments or room splash, at the same time checking to make sure that the room omni mikes being introduced didn't cause phase cancellation. In other words, we had to make sure that that room microphone was moved until its output was in-phase with that of the close mike picking up the original source.

Historical Sound Perspectives

The sound of radio broadcasts improved during the late Twenties, with the introduction of increasingly sensitive mikes — plus amplifiers and monitor loudspeakers — to take the place of the phonograph acoustic horns used for recordings. Guitarists Lonnie Johnson, Eddie Lang and pianist/cornettist Bix Beiderbecke took advantage of the development of such radio techniques, which were integrated into the recording industry during the mid-1920s. Possibly for the first time, these new microphones enabled all the beautiful subtleties of an acoustic guitar to be heard and captured on vinyl.

For our project, the blending of guitar sounds was easy, but capturing

Marshall stack for "Gypsy 68" — close mike is an EV RE-20, and distant a Neumann U87 (sometimes used instead of an AKG C452).

the cornet sound presented a problem. We placed cornettist Bill Berry in the dead side of the studio for isolation, and used a Coles 4038 ribbon microphone placed near the bell to capture the "blat" that a brass instrument creates as the sound comes out of the bell of the horn; EQ was restricted to a 4 dB boost at 12.8 kHz and +2 dB at 3.2 kHz. For the guitar parts I played an Ovation acoustic close-miked with an AKG C452; EQ was +6 dB at 12.8 kHz and +2 dB at 3.2 kHz. A Neumann U87 set to a figure-eight pattern, was used for the room sound and equalized the same as the C452.

The *Lonnie, Eddie and Bix* track was unique because here we are talking about two instruments that form a rather unlikely combination — guitar and cornet — and to get the two of them to "talk" together, to sound like you're sitting in the living room listening to these two guys playing their instruments, is an unusual request. In actual fact there are three instruments: cornet, plus rhythm and lead guitars.

On the track, *Swing Man Swing*, I wanted to feature the sound of a studio rhythm guitar combined with a jazz band. The two guitarists I wanted to characterize were Freddy Greene with the Count Basie Band, and studio player George Van Eps. Greene was known for his recorded work, and Van Eps for his radio broadcasts during the Thirties. Nick Ceroli was my choice for drums on the total project, and his drum kit provided a fantastic sound for big band jazz. I thought about recording his set with just two mikes, but changed my mind when I considered the sound control I would need in the final mix.

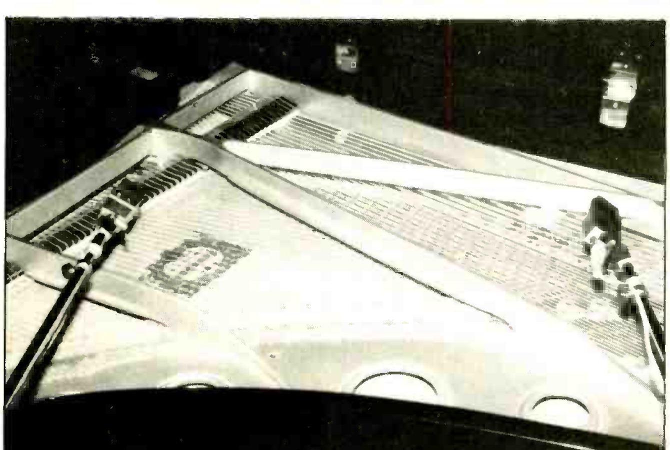
Microphone setup for the jazz drum sound I ended up using was as follows: kick drum — U47FET; snare top — SM57; snare bottom — Shure SM545; high-tom — C452 with 10 dB pad; low-tom — 452 with 10 dB pad; high-hat C452; overhead-left and right — C414s with 10 dB pad. A good jazz drummer is very smooth with his sound, and you are seldom surprised by sudden changes in dynamics.

The *Swing, Man, Swing* track represented a departure from the others on the album, because we employed more multitrack recording. It involved two guys playing cornet and sax, and then doubled; in effect we used multitrack to make it sound like a big band. I decided not to play on the basic tracks, but to add the rhythm guitar later. In this way, I could work as the producer, following my score and guiding the performance. We used one of my favorite microphones for guitar: an AKG C452 with a 10 dB pad.

Blues for Charlie is very similar to *Swing, Man, Swing*, since again the idea was to utilize multitrack recording to capture a big-band sound with very few players, by doubling or tripling horn parts. Charlie Christian used the full spectrum of music available to the electric guitar during the late Thirties and early Forties. Through recordings and radio broadcasts, he totally revolutionized the jazz guitar.

For an electric guitarist, understanding how to get the desired sound out of his amplifier is paramount. I use a card file system for each amp and control settings for each guitar.

Left: **The author's miking technique for Yamaha grand piano — a pair of AKG C414s to cover high and low strings.**
Right: **Mike evaluations for classical guitar included comparative listening tests between a C414, SM57, U87 and C452.**



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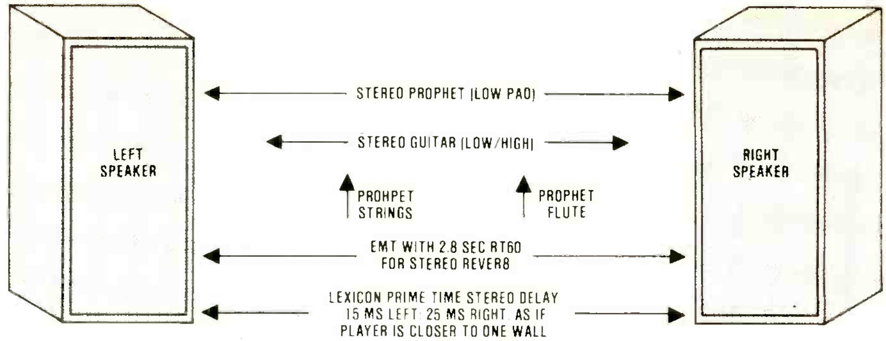
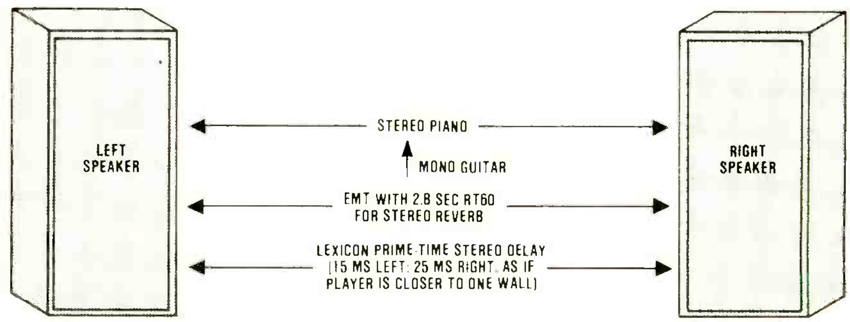


STEREO PERSPECTIVE AND PAN POSITIONS FOR "DREAMS" *(above)* **AND "A TUNE FOR BILL AND JIM"** *(below).*



GOING BEYOND NATURAL SOUND

The sound used for *Blues for Charlie* ran as follows:

Guitar = Gibson 175 with GHS strings medium gauge; front pick-up used only; tone control set at three o'clock relative to its off position.

Amplifier = Fender Princeton modified Phase II; volume 4.2, treble 6.8, bass 3.5, reverb 5.8, gain 3.6, and master set at 10.

For Charlie Christian's sound, and all other electric guitar sounds on this project, I used my personal-use studio to work on the textures I was after. Having a small collection of good microphones helps so, by the time I reach the studio, I'm ready. We used again the AKG C452 with a 10 dB pad to record *Blues for Charlie*.

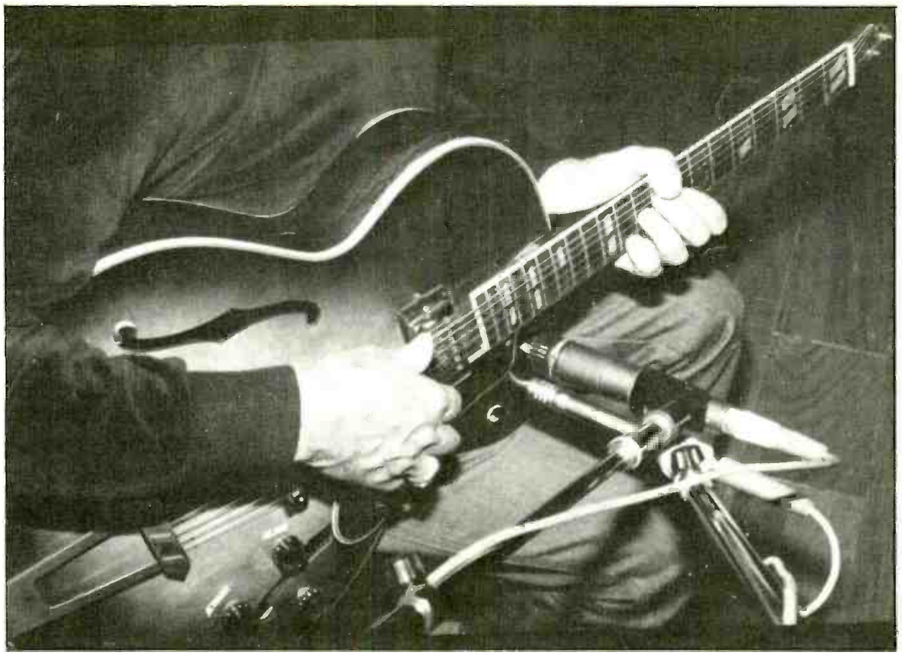
Fictional Collaborations

Gabor Szabo was one of the most recorded guitarists during the Sixties and early Seventies, and Carlos Santana was his biggest fan. The concept behind *Gypsy 68* was a fictional meeting of these two artists, Szabo and Santana, and tries to convey the way they might have performed together. I used an Ovation guitar with a DeArmond pick-up attached for the Szabo sound, and ran it through a Fender amp. Then we overdubbed the Santana sound using a Gibson Les Paul guitar with a stock Marshall four-speaker (Celestion) cabinet. This sound was also pushed by a 100-watt Marshall lead amplifier equipped with a pre-amp. For the final touches, the cabinet was miked with a RE-20 and a C452 with a 10 dB pad.

Trio was a tune that was written for Barney Kessel, a jazz guitarist who was hired for literally thousands of recording dates in Hollywood studios. Because this track used a guitar, piano and bass format, Joel DiBartolo's bass sound formed the foundation for the piece. We ran his guitar through an AXE direct box with a little bit of compression and no equalization. The guitar was miked with a C452 with a 10 dB pad and no EQ, and the piano microphones were two C414s.

Song for Carl and George was another recreation of two friends playing together, and communicating with their instruments. We tried to frame the dialog by putting them on opposite sides of the stereo soundfield, because the two artists are supposed to be sitting opposite one another, and communicating via the same piece of music. Carl Kress was one of the pioneers of the chord solo-style of jazz guitar during the time still considered the "Swing Era," and was one of the most successful radio guitarists in the Thirties. George M. Smith became the first staff guitarist at Fox and Paramount studios in Hollywood, playing all styles for motion-picture underscoring.

On guitar part #1, Carl Kress, I used a C452 for close miking and an omni



Guitar microphone technique for "Tune for Tal" — a Gibson 175 played acoustically and covered with a combination of a Shure SM57 and AKG C452 mounted close to the bottom of the fret board, to pick up the sounds of the low and high strings, respectively.

U87 for room ambience. The EQ necessary to catch this sound was +4 dB at 12.8 kHz, +2 dB at 3.2 kHz and -2 dB at 400 Hz, for both microphones. Part two, George M. Smith, incorporated a U87 omni pattern for the room mike with identical EQ, while the close microphone, one of my favorites, was a AKG C12, which we equalized +4 dB at 12.8 kHz and -2 dB at 400 Hz. Both parts were played on an Ovation acoustic guitar.

Jimmy Raney was one of the best jazz guitarists in New York, which city formed the cradle of jazz during the late Forties and early Fifties. Here he met and developed a musical rapport with tenor saxophonist Stan Getz, leading to many fine collaborations on record. I chose Hollywood jazz saxophonist, Don Menza, to play on this tune. His style and technique are flawless.

On *Jim's Tune*, in tribute to Raney, we were going for a very smooth sax sound, and ended up using a U87 for the solo. Because Don was playing so close to the mike, we used a windscreen to reduce some of the air that was pumping out of the instrument. We also used a little compression, and just a taste of EQ.

Tal Farlow received his greatest notoriety with vibist Red Novro's trio. Within this format — featuring Charles Mingus on bass — Farlow's guitar functioned as a solo instrument, an ensemble amenity, and a rhythmic bond. On a few of his recordings he played solo, unaccompanied acoustic guitar pieces.

Since I wanted to put the listener "inside" the instrument, on *Tune for Tal*, we employed an interesting guitar miking technique that involved a combination of a close-miked SM57 dynamic on the guitar's low strings, and a C452 condensor with a 10 dB pad on the treble strings. Both mikes were routed onto separate tracks, and during the mix we panned the low strings to the left and the high to the right. On top of this we added a little bit of digital delay to recreate a room sound, and reverb to give it some depth and duration; the result was a huge-sounding guitar.

Laurindo Almeida and Bolo Sete utilized nylon-string classical guitars in a fusion style of jazz with Brazilian flavors, which later became a standard by which the bossa nova style would be known. For the track *Batuque*, we used a guitar trio: classical guitar, bass, and drums. A Ramirez guitar was miked with a C452 mounted about 18 inches away to give the sound time to "develop" before it reached the microphone.

Classical guitar miking technique for "Dreams" and "Batuque," utilizing a combination of close-mounted and distant microphones to faithfully capture room ambience.

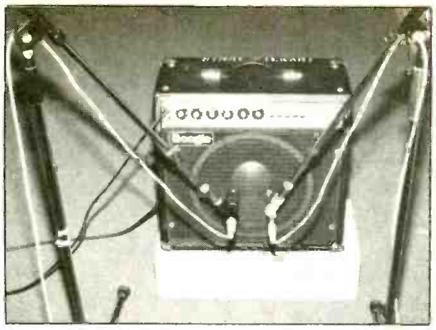
Dreams was a composition I decided to write in modest homage to myself. After playing on more than a thousand recordings, living through the changes in the recording process since 1955, and trying to make a go of it as a studio owner and artist . . . this one was for me. I chose a combination of classical acoustic guitar with synthesizer orchestra, in an attempt to relive those lush sessions we used to record live in the studios for records, and on soundstages for film and TV music.

On *Dreams*, instead of using just one mike and letting the sound develop in the room, we wanted to get a slightly different feel. An SM57 was used on the low strings, and a C414 on the high strings. Because we were close-miking the guitar, to overcome the bass promixity effect we rolled off a little bit of the bottom, and added a touch at the top. The second guitar track was recorded in mono with a C452 mounted 18 inches away.

Because of his musical ability and in-depth approach to his instrument, Jim Hall was destined to become a jazz guitar star. His duo recordings with pianist Bill Evans made Hall one of the foremost lyrical jazz players. I chose pianist Dave Benoit to work on the project, since on keyboard he relates those same qualities.

For *A Tune for Bill and Jim*, we used an AKG C452 with a 10 dB pad to cover the guitar. We added a touch of EQ: a little bit of high end (about 12 kHz and 8 kHz), and rolled off the bottom because the guitar produced such a big, fat sound. On piano — a 9-foot Yamaha Conservatory grand — we used a pair of C414s; one on the low and another on the high-end. Piano-mike placement for all of our tunes was done in such a way as to capture sound of the entire soundboard. However, rather than placing the microphones in such a way that they faced down at the board, we tilted the mikes off-axis a little bit to reduce the amount of acoustic phase cancellation that occurs when sound bounces off the soundboard. We also rolled off some of the extreme low frequencies, because there was so much thunderous bottom on this piano.

Acclaimed for his virtuoso technique, Joe Pass is a flawless player; his harmonic concept is impressionistic and his musical lines are long, somewhat similar to Bach's. An amplified Gibson 175 guitar was used for the track, *Joe's Soul-O*. Although a mike had been placed very close to the



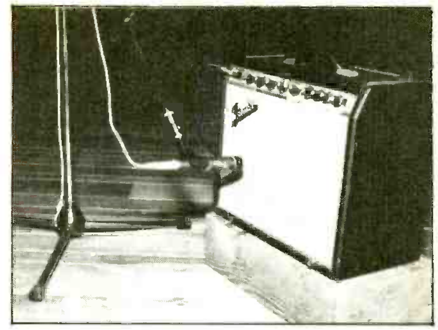
Miking of Mesa Boogie amplifier in hallway for "Wes" using a C452. (The SM57 served as an alternative mike on the session.)

backline amplifier, and another set up to capture the room sound, the output from both mikes were not combined on the same track. Instead, the two separate tracks enabled us to spread out the recording. We had the room mike panned over on one side, opposite the amplifier sound, to stretch it out a little bit and create a little bit of a delay from side to side.

While miking the amplifier cabinets, we placed the microphone in a variety of experimental positions to play with the overtone structure. We found that if the mike was placed in the center of the speaker, a very bright sound would result, but one that didn't have a very "full" sound. By moving its orientation outwards from the center to the side of the speaker cone, we were able to capture a warmer sound. Generally, at a point between the outer rim and the center of the cone, there's a place where we were able to capture the most overtones, and obtain a more accurate recording of the guitar.

One of the few jazz guitarists to become a household name during the Sixties was Wes Montgomery, who believed that music should be communicated to the audience. One of Montgomery's most identifiable characteristics was the sound he achieved by using his thumb instead of a plectrum. On the tune, *Wes*, I used a quartet setting: guitar, piano, bass and drums. To keep the guitar sound separated from the rest of the instrumentation, in the event I needed to touch up a few notes as overdubs, I placed the Mesa Boogie amp in the hallway of the studio and miked it with a C452 with a 10 dB pad.

The opening of the album's frontpiece is the sound of the Yamaha grand piano played backwards, which was achieved by flipping over the tape the wrong way, and having Dave Benoit play a long, sustaining chord. The attack of the chord, when the tape is replayed the right way, starts where the chord should *end*. (What Dave heard while overdubbing the



A Fender Twin guitar amplifier received a Shure SM57 to capture the Jeff Beck sound use on the album's opening track.

backwards piano was the synthesizerrs, which are already on tape, being played backwards from the end to the front of the song. When they got to the front, he hit his piano note, so that when the tape was flipped over the proper way, you hear the sustain first, the chord would end, and then the synthesizers would start.)

To achieve a Jeff Beck sound, I wanted to experiment with a few mikes. I was happy with the amp sound that I worked out in my personal-use studio, but something was lacking. In the studio, Chris and I compared an RE-20, a C452 with the 10 dB pad, and an SM57. We taped all three microphones, and he called me into the studio to listen. He had slated

CREATING AURAL IMAGES

Engineer Bruce Swedien's Varispeed and Reverb Techniques for Taking Sound Beyond Sound

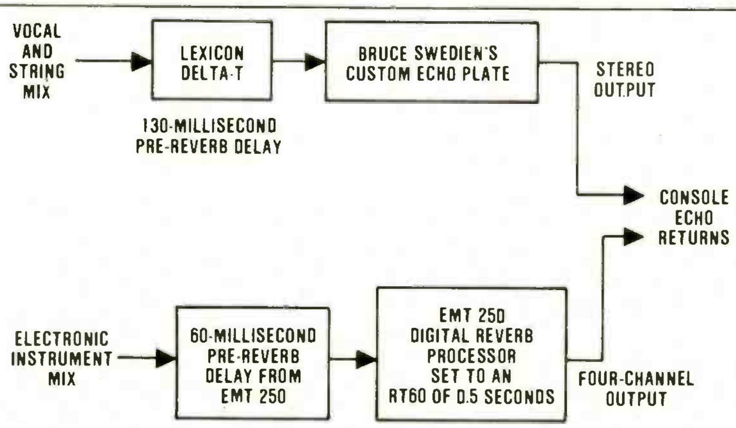
"Sound engineers must go out and hear live music in the environment where it is performed," Bruce Swedien offers, "Seeing and hearing classical music is more important than ever, with the increasing use of synthesizers that represent orchestral sounds.

"If there is a sound characteristic or tonal quality that is desired, it should be recorded during the performance at the tracking date. It's *very* difficult to recreate the unique special characteristic in the mix when you're dealing with a multitude of sounds.

"When working on sessions with Quincy Jones, he knows what he wants during the tracking session, and signal processing is not overdone; simplicity is generally the rule. The 'colors' are created by the instruments in combination with the right microphone and its placement; the purity of the sound is the *very first* consideration. With synthesizers and other electronic instruments, the player understands the variables and is allowed to manipulate the sound to get what Quincy wants.

"I varispeed an element to retain the natural sound instead of using a pitch-change device. Let's take, for example, a clave. I can retain the characteristics of the attack, but Quincy may want the pitch to be higher in the track. Then, VSO-ing allows me to retain the purity of the sound — the naturalness of the sound and attack — but I can change the pitch without changing the instrument's character. I am really going beyond natural sound. When this treatment of an element is then played back in the track at the correct speed, the attack and decay are only slightly altered.

"Your first consideration will be how far you want to take the sound. The more



To many engineers, Michael Jackson's "Billie Jean", from the *Thriller* album, represents a good example of the natural sound of the Eighties. Session engineer Bruce Swedien manufactured the characteristic sound by coupling existing, natural sounds with synthetic "ambiences," and a heavy beat. In the mix he creates a subtle blend by complementing these sounds with reverberation to provide space and depth.

Shown above is Swedien's reverberation setup for "Billie Jean". A mix of Jackson's vocal plus strings passed through a 130-millisecond delay from a Lexicon Delta-T DDL into his custom reverb plate, to provide a large hall sound with a rich depth of field. Electronic instruments were processed through a 60-millisecond delay into an EMT 250 digital reverb set to an RT60 of 0.5 seconds, to re-create a smaller environment. Outputs two and three from the Model 250 were used in a normal stereo configuration, while outputs one and four provided additional echo and reverb density.

Also used on the mix was an Inovonics Model 201 limiter-compressor to push up and emphasize the kick drum, while tom-tom fills passed through a dbx Model 160, followed by a Valley People Kepex II noise gate. The combination of 160 and Kepex II helped level the dynamics of this part, and brought out its presence in the track, Swedien recalls. On lead vocal a dbx Model 165 Over-Easy compressor was used to keep the dynamic peaks in perspective, followed by an Orban De-Esser to catch any sibilants that might otherwise have saturated the tape.

Sometimes Swedien will varispeed the multitrack to slightly detune a track, and which contributes to the stereo imaging when placed on one side of the stereo soundfield with the original on the other — this technique adds width to the stereo image, and offers a characteristic sound timbre.

GOING BEYOND NATURAL SOUND

on tape the different microphones, but did not tell me in which order they were recorded. Sometimes a musician that has had a lot of experience in the studio gets a certain idea in his head about the sound of mikes he is used to working with, and forgets to use his ears as a reflection — the final judge in choosing a sound. In this case, while listening to the playback of each, I heard the sound I wanted without knowing which microphone was used. To my surprise it was the SM57 that I liked, although I thought for sure I would have preferred the C452.

The Remix Stage

When I'm recording, the reverb chamber is one of my primary considerations. In tracking, even though usually I won't print reverb on tape, it is important for foldback and control-room listening. The EMT 140 plate used for tracking at Monterey Studios had a nice, bright sound to it. The reverb I wanted to use for the final mix, however, was at Sound City. Their EMT 140 sounds just great, plus they have an AMS RMX16 digital reverb that would give me the sound tapestry I needed for the final mix, which was to half-inch at 30 i.p.s.

I wanted to recreate concert hall-type echoes, bathing the listener in the sound; what I refer to as the "ear over a full stick." While each tune needed a different approach, our basic concept was to use a slap pre-chamber delay for guitar sounds; drums and bass in short echo; and horns and synthesizer in long echoes. Time-delay values to suit musical tempos were not worked out beforehand. (But I would refer the reader interested in calculating exact delay times to the article, "Creative Use of Delay Rhythms," by Roman Olearczuk; *R-e/p* February 1982 issue, page 68.) My method was not hit or miss, but by how it sounded in the mix as the delay related to a song's tempo. I would never mix a pop record this way, since the use of carefully worked out delay rhythms is critical in the production of such recordings.

In assembling the album, we used two techniques to sequence: a segue-fade-out at the end of a tune, by leaving the echo in and fading the musical information by five dB; and the other was the well-known butt splice. Two other techniques that I considered, but did not use, were the dead stop, reverse envelope and fading, using two tape machines to fade one tune into another.

We were trying to frame a jazz performance with the kind of clarity and definition that's available with current technology, and I think we succeeded. The miking techniques we used were pretty common; there's no magic involved. Mike placement has a lot to do with it, but getting a good performance from the musicians, to me, has everything to do with the final product. Also, once a musician feels confident that their work is going to be represented properly, they will give you that performance.

* * *

These and other ideas formed our concept for putting together the whole package. What we attempted to do was to capture something for the listener — you might call it "ear candy." My intention was to frame the instrument in such a way that it is presented to the listener as though he or she was actually part of the recording process. I tried to achieve a sound texture that is at once a performance, but also sounds as if it had been performed in a large hall.

Understanding natural sounds will become more important in the Eighties. Blending them with today's synthesized sounds will form the cornerstone for the production of great sounding records — the ability to go beyond natural sound. ■■■

CREATING AURAL IMAGES — continued . . .

variation of pitch that is used, the more 'synthetic' the sound will become. We will test out this process by experimenting with how far we can take it, and then listen back at normal speed until we find the sound we like. I have made up a chart which will translate VSO speed into musical terms — half step, quarter step, whole step, etc. We then know by looking at the chart how far, in musical terms, we will be altering the pitch of a musical instrument.

"A good example would be a percussion player with a collection of cowbells, all set up to different pitches. There might be one with very good clarity, but it doesn't have the pitch we want. This cowbell would be recorded at a VSO speed and adjusted. With Quincy's experience and knowledge as an orchestrator/arranger, he can really create in the studio; he has a great instinct for colors.

"I do a blend of taking sound beyond sound. By way of an example, consider a saxophone: its natural sound will be treated in a normal way, and at the same time I will mike the hook-up to a synthesizer. One track will have the classical sax sound, and on the other is the sax synthesizer element. Now I have the tonal characteristics and inflection from both sound sources — a process I call 'synthetic translation'. We now bring back this sound as a blend to create a balanced new sound — sound beyond sound; an Aural Image.

"The type of music determines its acoustic environment, which can involve echo or a blend of echoes into reverberation. This can be recorded in the studio, or an image created in the mix with acoustical support from echoes and reverberation. By listening to live music in its acoustical environment you build your own benchmark. An engineer needs to develop his or her own set of ears and creative personality. To take a recording and create something new and different by using all the means available to us, that's when the musical personality of the engineer comes out.

"Consider another example: on an R&B vocal, DDL and the shadow of a single echo on the vocal could be used to trail the vocal. I figure out the beats per minute in the track and create a whole-note, half-note, or quarter-note echo, as it relates to the tempo. I use a digital metronome to tell me how many milliseconds these note values will be. Sometimes I mike together these echoes. I use mike placement to create an effect, one mike away with rich delays and one with no delays, and spread these left and right for the same sound source." □□□

ALPHA STUDIOS

MAKING THE TRANSITION FROM RECORDING STUDIO TO FULL-SERVICE AUDIO-FOR-VIDEO AND FILM RE-RECORDING FACILITY

by Ralph Jones

Photography by Kathy Cotter

Many studio owners must wonder what the future holds for the recording industry, which currently seems to be experiencing a time of tremendous flux. On the one hand, there is a clear excitement about the music we are all involved with: A new music video show seemingly appears on television and cable practically every week, and we are all aware of the recent phenomenal successes of several relatively low-budget feature films that utilized strong music and dance elements. On the other hand, with one or two notable exceptions, record sales have not really climbed to the heights that one might expect and there are many economic pressures bearing on commercial recording studios.

It would appear that the public's buying patterns are changing. Given that the present forms of distribution for music increasingly are swinging towards the visual media — or at least that a visual component is becoming increasingly important to the marketing of music — it's only natural that a growing number of studios are looking to accommodate the particular needs of these new media. And its a trend that is being accelerated by the current (and long-overdue) advances in the quality of film sound — one prominent example being the Lucasfilm THX theater sound system — as well as recent developments in stereo sound for television broadcast.

Alpha Studios owner and engineer Gary Brandt is an industry mover who is making a strong play for the visual media markets, and his case is instructive in many ways. Brandt's first facility — which he now refers to as Alpha One — was profiled in the August 1981 issue of *R-e/p*. Alpha One was remarkable as an uncommonly small facility — the recording area measured 15- by 12-feet — that had turned out a series of hit recordings, including Robbie Duprees' *Street Corner Heroes* and *Robbie Dupree* albums. The new Alpha Studios, built on the site of Leon Russell's ill-fated Paradise Studios, is somewhat larger — counting soundlock, studio, and control room, the new facility occupies about four times the space of Alpha One. Furthermore, in contrast to the older facility, the new studios are equipped to serve broadcast video post-production requirements, and also includes a large soundstage area with full lighting grid.

While Alpha Studios looks more like a world-class 24-track audio recording studio than a media post-production house, it does in fact pride itself on being a full-service facility that can handle any form of audio conforming within the one-inch videotape broadcast standard, including scoring with large ensembles of musicians, automated dialogue replacement (ADR), Foley, and sound effects recording. Brandt has two feature cartoon shows as clients already —*Magnos* (which has now been syndicated) and *Andromeda Stories* — and will be doing two more feature-length animated films during the next three months. All of the projects are single-cell animated shows purchased from Japan by International Media Corporation.

Alpha handles all of the sound elements for these cartoon shows, using SMPTE-locked videotape transfers from the original 16mm film, and audio transfers from 35mm mag to a half-inch four-track format for both music and effects, laying the audio back in the appropriate spots after the product is edited. Included in the project are dialog, music and sound effects replacement, all of which is achieved without the use of prerecorded NAB cartridges and cart machines; among other endeavors, Alpha currently is developing a new standard for event control based around the BTX Cypher Code System controlling multitrack transports and one-inch videotape machines. Audio for the shows is assembled on a 16- or

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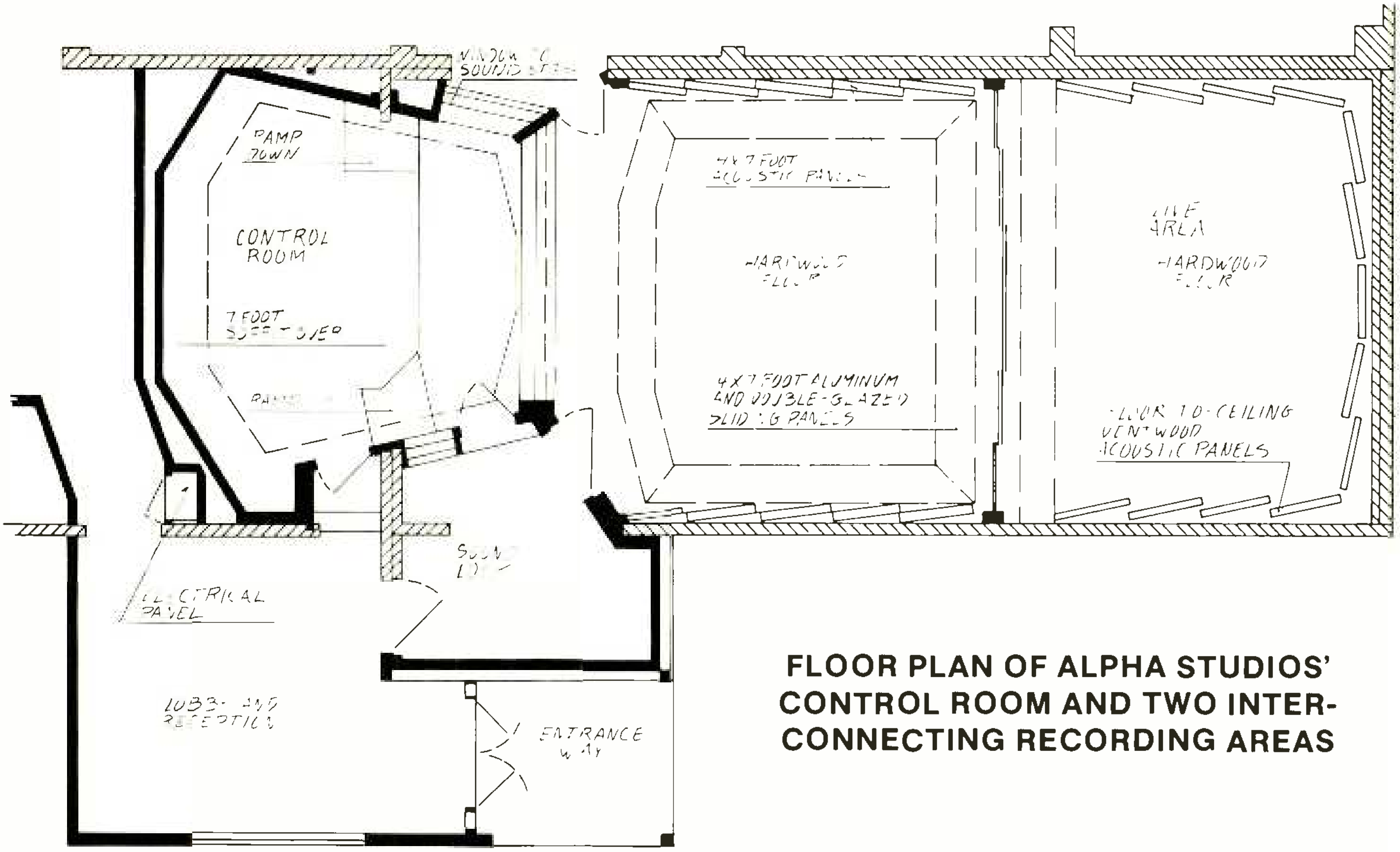


FLOOR PLAN OF ALPHA STUDIOS' CONTROL ROOM AND TWO INTER-CONNECTING RECORDING AREAS

ALPHA STUDIO

24-track machine, and the final mix bumped back over to one-inch.

Studio Design and Construction

Ground was broken for the new facility in January of this year, and celebrated its official opening August 22nd. What state was the building in when Brandt took possession, we queried?

"This was a shell when we acquired it," he recalls. "Leon Russell never had a studio here. He had a remote truck and a video truck housing all his equipment; there were big multiway snakes running through holes in the wall, and the rest was basically a warehouse. He had spent some money on the electrical system; we walked into easily $150,000 in outlets and wiring just sitting here. There's a separate isolation transformer, and a separate subservice for the building. We started with an 800-amp panel, and we were told that we could easily go to three or four times that with the subservice that we have. So, in the soundstage area, we could fire up arc lights if we wanted to. We have the possibility of expansion in many directions."

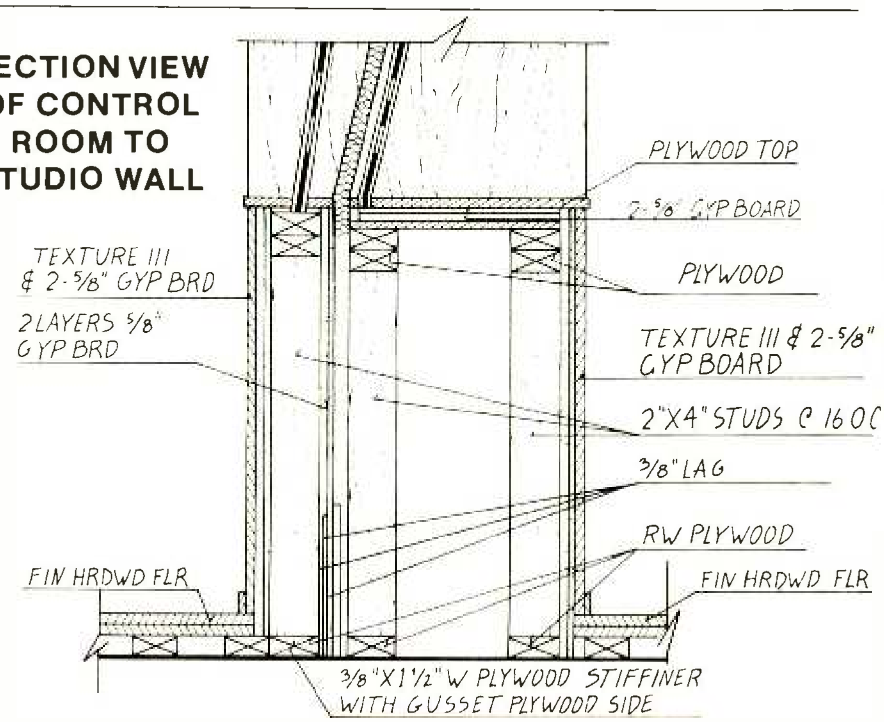
It goes without saying, of course, that Brandt and his partners have also spent some money of their own on the new facility. The studio, like the control room and client lounge, is finished predominantly in Ventwood, a commercial product that comprises one- by two-inch Redwood slats dowled ¾-inch apart.

Alpha's owners have constructed a room that features a live acoustic design. "We wanted the old sound that all the good sounding hits were made of when we were in high school," Brandt offers. "Most studios have a dead sounding room, and we've been dealing with dead rooms for a long time. The live sound is today's sound."

For his former facility, in an attempt to conserve space, Brandt had developed a construction method that involved turning the studs to the room, and putting treatment materials on the back side. Structural members of the room thereby became a part of the acoustical treatment. The same method was followed in the con-

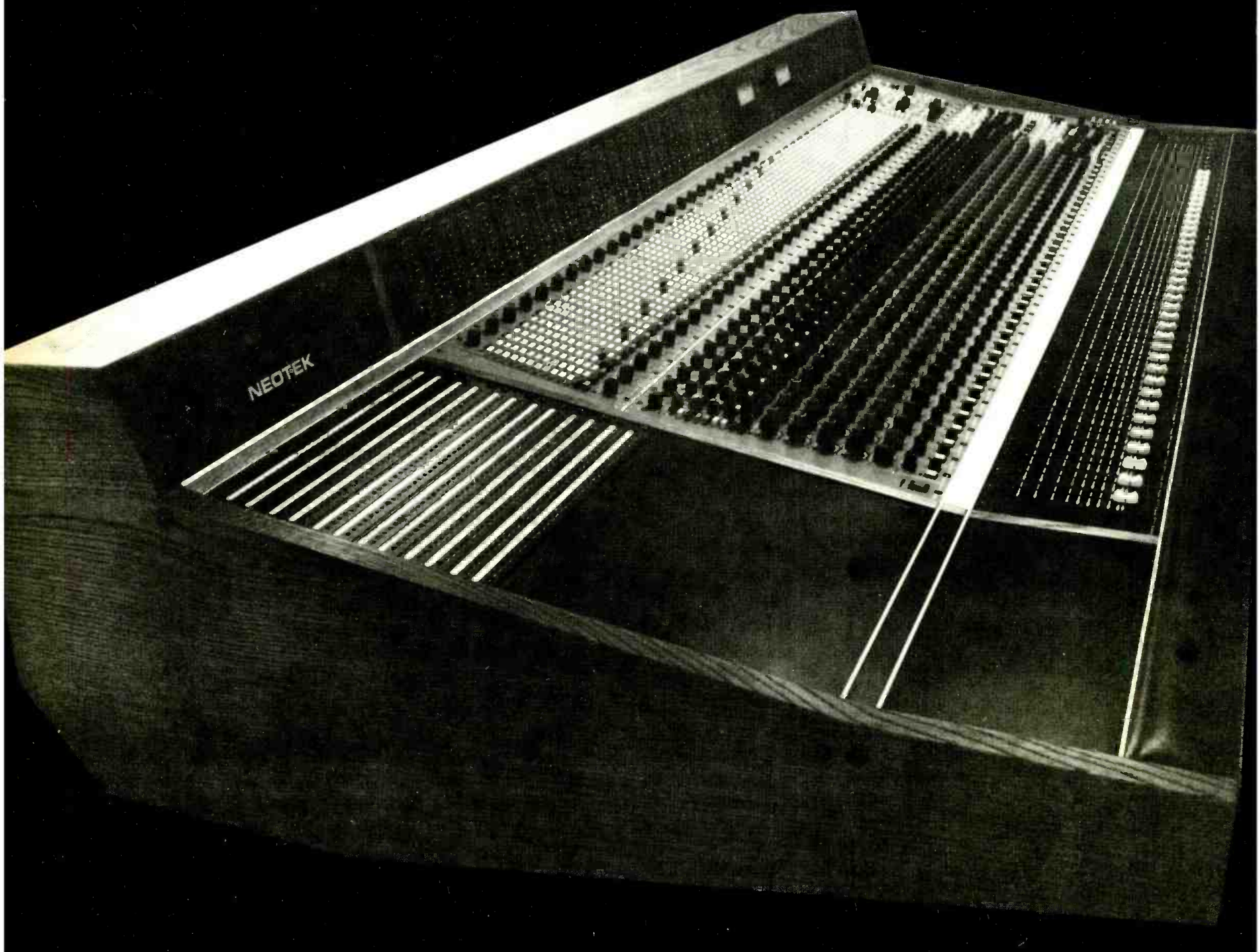
— continued on page 108 —

SECTION VIEW OF CONTROL ROOM TO STUDIO WALL



It's not only

what we make,



important for multimedia clients, Brandt says, who require submixes into many formats at one time. During film mixing, for example, you need to remix several elements simultaneously; as a result, most film re-recording houses normally feature 60- or 70-input consoles manned by separate music, effects and dialog engineers. "That's a totally different direction than we've gone in," Brandt says. "With our 32-input console we are able to do exactly what the video industry needs, and certainly most of the prep work for film. We can also use the foldback section to provide a total of 66 inputs during complex mixes.

We're not using a 35mm mag standard here; instead we're using 24-track master recording. We're doing what all the other studios are going to end up wanting to do."

Alpha Studios' control room features a customized API console, which has been extensively modified for multimedia video and film re-recording. A total of 66 inputs are available during mixdown, routing to 32 discrete output busses.

An essential (and unusual) element of this studio is its videotape machines. Built into a custom alcove in the control room's rear wall are two Ampex VPR-2 one-inch transports; the alcove has its own lighting and plex doors to eliminate machine noise. Video patching has been provided for all half-inch VHS/Beta and ¾-inch U-Matic copy machines, along with DAs for video and sync distribution. The control room has video tie lines that will eventually terminate in a video control area for the stage, but whose function for the moment has been left open. The tie lines could serve any number of uses, including film or video edit control.

The studio also has a video test set, an absolute necessity, Brandt considers, not only for maintenance but also to provide a stable source of house sync. Video and audio transports are SMPTE-locked using the BTX Shadow system, complete with the new Softouch Controller (described in greater detail in an accompanying sidebar).

What outboard equipment has Alpha added to meet the particular demands of multimedia work, we asked? Without hesitation Brandt came up with the answer: stereo limiters. The studio has two UREI 1178ST limiters, that are used constantly, the engineer says, since submixing in post production very often requires stereo limiting of music and effects.

Live-Performance Soundstage

The stage area features a 1,000-amp power grid, separately branched into 105 circuits, with a lighting booth above. Lights can be gripped at the ceiling (12 feet), which provides about two feet to play with above a standard 10-foot set.

"That's just enough to do it," Brandt concedes. "This is an old building; the floor doesn't have any trusses, so all we would have to do to get more height is go straight down and underpin the footings on the front exterior. We could go down six feet and build a ramp up, but since that would cost $30,000, we're not jumping on it.

"We want to have that room in limbo for about a year and a half or two years, to evaluate the state of the business." Accordingly, so far no acoustic treatment has been done to this area.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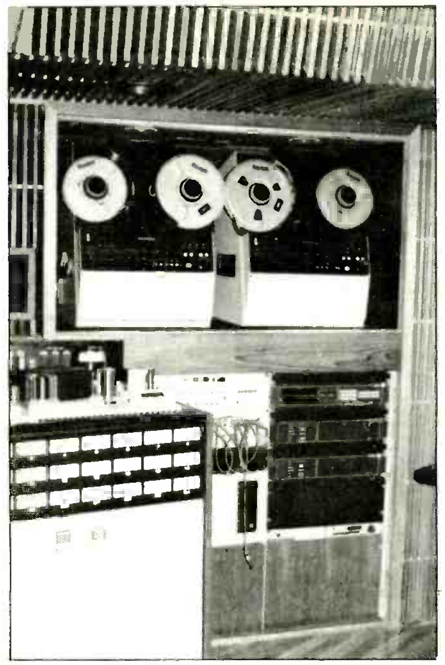
Audio for Video

The medium of television has not been known for placing an emphasis on quality audio; sound circuits in domestic receivers normally are limited in bandwidth, and speakers are of poor quality. Yet at Alpha Studios there is a clear emphasis on quality: upon walking through the control-room door, one encounters a true world-class tracking room, *not* a dubbing stage. What's the point you might be asking?

"Stereo broadcast is just around the corner," Brandt reflects. "There is definitely a coming revolution in the quality of TV audio, and the people who are not prepared to do high-quality sound for these media are not going to be working.

"The real uniqueness of our facility, is that our background in audio is 15 years deep — both in engineering and technical backup. We can make a better recording of a musical work than the guy down the street who does audio post all day. We can get the client into special effects work, character treatment of dialog using synthesizers, and so on. We're bringing high-quality sound to the video and film industry."

Little wonder that Gary Brandt takes such a stance. His engineering credits include a Prince album, Robbie Dupree's "Steal Away" and "Hot Rod Hearts," and "Blame It On The Night" with Terry Williams. Poco, Michael McDonald, The Doobie Brothers, Seals and Crofts, Delaney and Bonnie, Emmylou Harris, and many more artists have passed through his studio.



A rear-wall soffit houses a pair of Ampex VPR-2 one-inch, C-Format VTRs, for audio layoff and layback, and transfer to U-Matic and Betamax videocassette work tapes.

We asked him about the task of mixing sound to be reproduced through television loudspeakers: What must be done to make a mix that will serve for pressing, but still translate well in a medium for which band-limited mono is the standard?

"I find that trying to get the standard of your balances to the point where it's as linear as possible and as equally balanced from low frequencies to high frequencies — against a flat standard — is *the* most important thing," he responds. "The last mix I did was for Stevie Goodman, who wrote Arlo Guthrie's 'City of New Orleans.' Bernie Grundman just mastered it for me, and he said that side B required only 1 dB of equalization on one or two of the cuts. Basically, it was a flat transfer to the disk.

"That's what you're going for; not a sound in particular, but a *musical* entity. What parts or notes are best for the song? Where do they fit? Those are the decisions that are going to make the hits, not whether or not there's 2 dB of EQ on one thing or another. It's a musical *not* a technical decision."

There are, of course, new technical considerations to grapple with when working with sound locked to a video or film image. The SMPTE timecode track must be continuous and consistent for all the reels — dialog, effects, and music — of a show. Problems can easily occur if planning is not done in pre-production to assure that the code tracks are consistent, and they must be retained intact during the editing process. Engineers and producers who contemplate entering the visual media markets must be aware of such considerations, or face horrendous synchronization problems that require large, expensive blocks of time to solve.

A recent experience at Alpha is a case in point. "We had a show come in called 'Bravissimo'," Brandt recalls. "It was a Latin concert for the ABC Television Network. They start mixing, and of course things are supposed to go perfectly smoothly. But they quickly realize that the audio was going out of sync with the picture. They start looking at me and saying, 'What's wrong with your equipment?' To which I replied: 'Well, it's real simple. You've got an edit there.' They were real *obvious* edits, too, so they couldn't flag that one away from me. 'It looks to me like you dropped two frames on the edit,' I said, 'Here it's two frames off, and all of a sudden the timbale comes down a beat late.' Things like that started happening all through the show.

"Somebody in the production or artistic element had decided to cut the show up in various pieces. And we realized that some of their audio reels didn't have drop-frame timecode like the master did.

"Now, there's absolutely *no* way to conform a non-drop-frame source to a drop-frame, and keep it in sync, short of playing all sorts of manipulating games of pushing this machine off into a slow-lock condition so it doesn't drift as fast.

"With the Softouch, you can do about as well as you can with any other box by just playing around with it, but it doesn't work; no synchronizer can handle mixed code. Somewhere along the line, the show's going to be out of sync, and you'll start to notice it.

"We basically took the show and just did offsets and punches all through the tape to finish it up. It took an extra 12 hours of edit and audio-post time to clean up the show and get it in frame. When we were all done, there was one little dialog section that was about a frame and a half out, and you couldn't really tell. I don't speak Spanish, so of course I couldn't really tell how close it is, but it looked pretty cool. The show was conformed and fixed, but the engineer who brought it in had no idea how to fix it when he walked in the door. I sent them a bill for the extra time, and they never paid it.

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CUSTOM SYNCHRONIZATION SEQUENCES FOR AUDIO-FOR-MULTIMEDIA PRODUCTION

Hands-on Experience with the Programmable BTX Softouch SMPTE Synchronization and Editing System

The single most important component in any multimedia post-production facility is a timecode synchronization system. Apart from ensuring that all the audio and video transports being used in a session are kept locked together, a synchronization system allows an engineer to assemble the sometimes bewildering array of individual tracks that form the various sound elements of a multimedia production.

Several synchronizing systems, varying in degrees of sophistication, are currently available to studios, with several code standards in use. Alpha Studios' owner, Gary Brandt, chose for his new facility the BTX Shadow II, a modular synchronizing system available in a number of configurations, linked to a Softouch Audio Editing System. The Softouch is an intelligent programmable control console that incorporates both dedicated function keys, and a bank of "Softkeys." These are keys whose function is not defined, but which call (and execute) from internal memory registers various sequences of user-programmable key strokes. It is this programmability which, according to Brandt, makes the BTX System "the most powerful lock-up system you can buy." Eight Softkeys are provided, each of which has two registers for an effective total of 16 programmable keys.

"You can do a lot of useful things with Softouch," Brandt says. "Say I'm doing an ADR session, and they blew it in the dialog. I go back, stop the machine, roll back a little bit on the video machine, hit one button, and it automatically cues the machine to that point, goes back against SMPTE, and punches in from that point. That's my 'Punch-in Cue Point' program [see listing below] which is the kind of thing you use on every session."

Without the programmability of the Softouch, the engineer offers, that seemingly

"Those things happen in the industry. Sometimes the producers don't understand why there is a problem. Unless they have a deep understanding of what happens in the process, they're lost."

Digital Audio and Multimedia

Given that Alpha's console has been modified for 32-channel monitoring, in anticipation of digital technology, does Brandt have plans to make a move towards a multitrack? His answer may provide a hint or two for the manufacturers involved in this new technology: "Right now, in all multimedia work, crosstalk is a problem. If you mix to four-track, you can only use tracks 1 and 2; track #3 is too close to the SMPTE track #4. In 24-track, a client who wants to save money on the tape will run 15 ips, and split the tape, by running 12 tracks on the top and 12 on the bottom, and putting timecode on tracks 1 and 24. You're limited as far as what you can do with those outside tracks. Digital makes a lot of sense for those clients."

Brandt's evaluation seems to provide a different emphasis to the one we usually hear in discussions of digital recording, which normally stresses dynamic range, noise, linear

. . . continued overleaf —

CUSTOM SYNCHRONIZATION — continued . . .

simple operation would require dozens of keystrokes, and an inordinate amount of session time.

The real advantages of programmable synchronization systems appear when they are asked to do tasks that are not routine; it's the extraordinary case that proves the point. Brandt says that is the most sophisticated Softouch program, although he doesn't use it on every session, and saves real money when it is needed.

"We needed a sound effect of boat-harbor sounds on a master reel. We only had 10 seconds of the effects, and needed 60. I had developed a program that loops six times; I give it a mark in at the beginning, a mark out at the end, press one button, and it loops the effect six times onto my master reel.

"Think about it: The program is off-setting the master machine by exactly the

Studio owner Gary Brandt at the BTX Softouch SMPTE Controller.

amount necessary to extend that 10-second piece to 60 seconds with *no* editing of tape. It would need a 150-keystroke entry on any conventional lock-up system to achieve it. Because of a user-programmable Softkey, you press one button and go get a cup of coffee."

In any complex system, however, there is room for improvement, and being well-versed in the operation of Softouch, Gary has suggestions for system enhancements. "I happen to have programs that work well on the Softouch SMPTE controller; I know what I want the machine to do, and what I want it to do in the future. I would like to see BTX become more aware of what the consumers of these items are concerned about."

One such suggestion involves interfacing the punch-in and punch-out functions of Alpha's Ampex VPR-2 one-inch C-Format video recorder. "In terms of production, slaving that machine is real important," Brandt offers. "If you have a client who's just finished his CMX edit decision list, and he finds that the sound of a music or dialog piece is bad, he's got to go back into the post house with its big, expensive CMX editor and say: 'I need you to move my source-reel sound back over and clean it up. It doesn't sound right, and I need to conform it again.' Constantly, this is a problem in the industry.

"We have two VPR-2 machines with perfectly aligned audio, and we are capable of moving that audio back into any locked, offset condition to the one-inch master, punching in and out on cue. It's a valuable ability that forms a big part of our business. We don't need the CMX editor to do it. We were hoping that we would be able to conform the one-inch machines with a simple Softouch sequence. We've got it working, at least, thanks to my studio technician, Denny Shaw, who has done most of the work necessary to get the BTX system to slave the Ampex videodeck."

Brandt has a wealth of suggestions for capabilities he would like to see incorporated into a SMPTE-based synchronization system, all of which were born of practical experience in the highly competitive market he has decided to brave. A detailed treatment of these ideas is well beyond the scope of this article, and would contain information of limited interest to those not intimately involved with timecode synchronization technologies. It is apparent, however, that the users of these systems can provide manufacturers with valuable feedback that should be of particular use in the development of the latest-generation hardware.

For the benefit of *R-e/p* readers that are using a BTX Softouch controller, included below are listing of three programs that have been developed by Gary Brandt; all the programs assume that the record machine is configured as the Slave:

Program #1 — Loop at Cue Point

This program causes the system to punch-in at a cue point defined by the user; punch-out is manual:

SLAVE ASSIGN	MARK OUT
CHASE	STORE
LOOP RECALL	RECORD OUT
00	LOOP BEGIN
MASTER STORE	ENABLE
RECORD IN	SLAVE ASSIGN
RECALL	ENABLE
RECORD IN	MASTER STORE
STORE	RECORD OUT
MARK OUT	RECALL
1	RECORD OUT
00 00 00	STORE
TRIM	MARK OUT
MARK OUT	SK 9
RECALL	

(Optional: store an identification message in Softkey #9 if desired; otherwise, omit this instruction.)

Program #2 — Record Out At Cue Point

This program causes the system to punch-out at a cue point defined by the user; punch-in is manual:

CHASE	RECORD OUT
SLAVE ASSIGN	STORE
LOOP RECALL	MARK IN
00	1
MASTER STORE	0
RECORD OUT	0
RECALL	0

(The above four steps set the system up for 10 seconds of preroll to cue; alter these values to set a longer or shorter preroll interval.)

TRIM	LOOP BEGIN
MARK IN	SLAVE ASSIGN
ENABLE RECORD	SLAVE ASSIGN
SLAVE ASSIGN	SK 10

(Optional: identification label — see #1 above.)

Program #3 — Layover EFX With Offsets

The user manually cues the master and effects machines to the desire edit points, hits the Softkey in which this program is stored, and the system memorizes the offsets, prerolls, and automatically dumps the effect onto the master:

SLAVE ASSIGN	OFFSET
LOOP RECALL	23 HOURS
00	STORE
MASTER STORE	MARK OUT
MARK IN	FOLLOW
RECORD IN	LOOP BEGIN
OFFSET	ENABLE
AUXILLIARY STORE	SLAVE ASSIGN

□□□

ALPHA STUDIO

frequency response, and so on. What about the *sound* quality of digital systems?

"I think that the people who make digital machines should try at least to pass a reasonable-looking 1 kHz squarewave before they consider themselves better than analog," Brandt answers. "Cart machines will make reasonable-looking squarewaves, and here are these digital machines that can't. You can't tell me that doesn't make your music sound harsh."

Strong words, maybe, but the opinion being expressed is an important one. Clearly, even for the multimedia market, it might be necessary to re-evaluate sampling standards, and to deal with the phase-domain response of digital systems.

Planning for the Future

Alpha Studios is starting its business with audio-for-video post production, but that is definitely not all there is to the visual media markets. The creative flexibility offered by audio multitrack recording equipment locked via timecode to video playback may also be harnessed for film production, and can result in an increased efficiency of post production with greatly enhanced special-effect capability. A final cut of a feature, transferred to video with timecode, can easily serve as the reference for scoring and dubbing sessions, enabling the creative element of the project to take advantage of the enormous sound-manipulation capabilities the recording industry has at its command. The completed audio tracks may then simply be bumped back to mag film. Such multitrack technology is just beginning to be exploited by the film industry. (George Lucas and his staff at Lucasfilm, for example, are prominent pioneers of such new techniques, and Disney Studios also dub from code-striped video.)

We asked Brandt about such uses for his facility, and it turned out that he already had some experience with the technique, having used it to record a musical piece for a recent John Cassavettes project for Cannon Films. "It's a wonderful idea," he continued. "And we want to get more film people into that element. We're going to have a lot more work as a result, and our clients are going to find out how wonderful it is to work like that. It's so much quicker to work with multitrack than having a guy there singing with a mag machine. That's so 'Old School'.

"We are very interested in film. For the equivalent number of pixel points

An unusual feature of the control room acoustics is the pyramid-shaped ceiling, designed to reduce standing waves.

on a 16mm neg, with the best film available, you would have need 1,250 horizontal lines per inch to accomplish the same resolution in video. The very best high-resolution video cameras can only provide 1,000- to 1,200-lines resolution. And 35mm would require 2,500 lines per inch; 70mm 5,000 lines.

"Eighty lines of resolution is equivalent to a 1 MHz bandwidth. The best video recorders can coax 10 MHz on tape, which is equal to 800 lines of resolution; 16mm already has blown that away!

"I think film will be around for at least the next 10 years, because video can't duplicate the definition of film. The large screen is still an exciting medium for features.

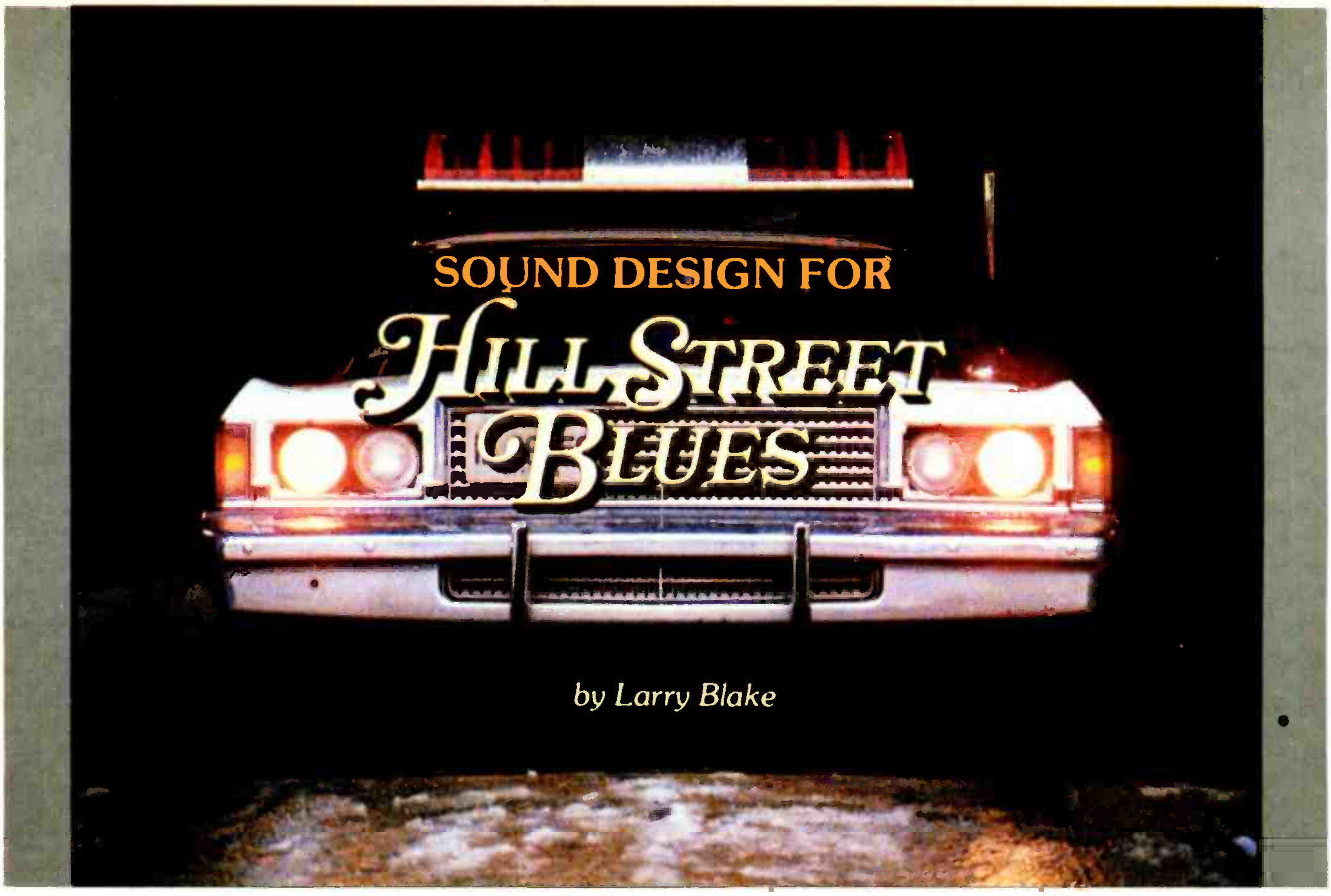
"The other parameter of film that interests me is this: If you were to take a frame of film and expose it to a setting where you have a very light subject in one corner and a very dark subject in another, the lighting contrasts obtainable are a 1,000,000:1 — an unlimited contrast ratio. In video, you can get maybe 100:1, if you're lucky; generally, they run 30:1. So we're not seeing the real dynamic effects of lighting that you're going to see on the large screen.

"That's where the state-of-the-art-of-video sits today, and why I've made a decision to look into available SMPTE-based film editing equipment, and go for a film-editing lab instead of video editing. We have video machines, because we would like to do tranfers."

Again, Brandt stressed the importance of good planning in preproduction when using timecode synchronization systems. "An engineer friend of mine told me about a film project he was involved in at [a Hollywood] studio. They had drop-frame code and non-drop code; they had 24-frame code, and they had 30-frame code. They had *all* the problems to deal with, only because they didn't take a hard look at what they were undertaking when they started. It's not that difficult, and I hope that events like that don't scare the film people away from doing something that's easy if it's done correctly."

* * *

Clearly, the visual media provide several fertile new fields for studios to work in, if they are able to make the necessary equipment investment. The present growth in sound quality in feature films, and the hoped-for arrival of Stereo Television, provide new opportunities for our industry. There seems no doubt that post production in these media will be a very different proposition in the future than it is at present, and the expertise that the recording industry can offer puts us in an ideal position to reap benefits while leading in the evolution of sound for the visual media. ■■■



SOUND DESIGN FOR HILL STREET BLUES

by Larry Blake

Very few television shows have attracted as passionate a following as NBC's weekly prime-time series, *Hill Street Blues*. With each episode covering a day in the life of an urban police precinct located in an unnamed large city, viewers are treated not only to an unsentimental look at life on both sides of the law, but also to all the grey areas inbetween.

The sound for the show receives a great degree of attention from its producers, a rarity in the tightly scheduled and budgeted world of episodic prime-time television. Beginning with the series pilot, which first aired in January 1981, sound design for *Hill Street* has been overseen by co-executive producer Gregory Hoblit. Because Hoblit is tied up with administrative chores, in addition to directing many episodes, the day-to-day post-production supervision has become the responsibility of associate producer Ellen Pressman. This article will take a close look at the philosophy behind the sound design for *Hill Street Blues*, and how Hoblit and Pressman provide a team of sound editors and mixers with the opportunity to push television sound to its creative limits.

Sound Philosophy: Noise as Texture

After reading the script for the series pilot, written by *Hill Street Blues* co-creators Michael Kozoll and Steven Bochco, Hoblit recalls that his concept of the sound design became self evident. "I had spent years knowing what I liked, and what I didn't like," he says. "And knowing what I would do when I got the chance. The minute I read the script, which had an overwhelming visual and auditory sense to it, it just jumped at me.

"We began with an overall notion of *noise*, whether it was visual noise, bodies constantly in motion, crossing each other; before you're finished with someone [on camera], someone else enters the scene. The idea was that noise was a constant, woven texturing device.

"I suggested to Billy Cronjager, the DP [director of photography] for the pilot and the show's first year and a half, that the film should look as if you took the negative in the morning, stretched it between the stages, ran a truck over it, and then stuck it in the camera. I wanted that heavily *degraded* feeling to it.

"In dealing with the production mixers it was the same thing: I wanted ambient noises on the track.

"Wardrobe was messy; people and cars weren't neat and tidy. Whether it was set dressing or wardrobe, like the world it was not perfect. Television tends to sanitize things, with all of the buttons in the right place and people speaking in clearly articulated sentences. On *Hill Street* there was an *intended* mess. Sound played a huge function in this, because pictures tend to be clean, no matter what you do.[1]"

Production Recording: As Realistic as Possible

It is the shared opinion of many mixers currently working in Hollywood that the "loop-it-later" syndrome of dialog replacement originally began in television production during the Sixties. Directors that would not be present at post production found those three words easy to justify, since waiting for the sound mixer took precious time from his short shooting schedule. The director most probably was not around to find out that either: *A*, the scene was *not* looped, and the re-recording mixers had to break their backs to make the dialog work; or that *B*, looping can often destroy an actor's performance. Later, these same directors moved onto features, taking this "fix-it-in-the-mix" attitude with them.

Although *Hill Street* shares the same seven-day shooting schedule as all other hour-long TV series, the approach towards production sound is fundamentally different from many television shows (and feature films, for that matter) in two respects. First, there is almost total reliance on the production track; only a few lines per *season* are looped. Which approach is in contrast to a unnamed major studio that sends actors to a looping stage every day after they are finished

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Hill Street Blues

shooting, and has them read wild (i.e., not to picture, which obviously hasn't even been developed yet) to quarter-inch tape, in anticipation of replacing the production track. (The idea being, of course, that the meter is running at a very fast rate during shooting, and big bucks can be saved if sound is never the reason to do another take.)

Therefore, the ability of production-sound people to deliver useable tracks is very much a function of the attitudes of the producers, and on *Hill Street* this is the least of a mixer's problems. Hoblit notes that "most shows shoot the schedule; our tendency is to shoot the *script*, thereby costing us in terms of the shooting schedule, which of course translates into dollars. There is a constant tug of war involved in doing it that way, but it has resulted in the type of show we have.*

"I know that the DPs could light [a scene] a third faster if they just flat lit it and 'phoned it in. But we shoot the script, and this attitude extends to casting, to performance, to writing, to sound mixing . . . to everything. We pay a painful and expensive price; one which can't always be justified.

"The toughest nut to crack is to 'paint with the soundtrack.' The talented people [in film sound] are few and far between, and most of them are out there doing features. It's one thing just to stick your mike in front of someone and work the dials, but another to do it artfully. It's a bitch to do in the course of shooting for seven days, 12 hours a day. But we do what we can so that when Ellen goes on the [dubbing] stage she is not trying to correct our mistakes. I'd rather pay the two bucks here than the four bucks there. We are saved constantly in post production, but there is the *intent* to get it all up front. If it's looped, you can tell the difference."

The second way in which the attitude towards sound for *Hill Street* differs from many TV shows is the infrequent use of radio microphones. (They were employed fairly extensively, however, on early episodes of the show.)

"In the hands of very good people," says Hoblit, "radio mikes are pretty effective pieces of business. However, they tend to have that 'tinny' sound, and to be a little thin. Actors on the show didn't like them, and some were really *hostile* to them. Also, they take time: radio mikes inevitably screw up somewhere and you have the poor mixer running from the panel saying 'Wait a minute!', and working with the antennae. So there is a combination of factors that mitigate against using them, as far as I am concerned."

Bill Marky served as *Hill Street's* production mixer for two and a half seasons, during which time he only used radio mikes twice. In those two instances, his boom microphone was physically forced out of the frame because of camera position. Whenever Marky was forced to use radio mikes, he will " 'dress' [blend the sound of] the radio into an open mike as soon as I can get the boom in there. Then, as the actors start to fill the screen, I will end up on an open mike."

Marky's favorite boom mike throughout the years has been the Sennheiser MKH816-TF. (The two-letter suffix, TF, indicates that there is no mid-

*According to associate producer Ellen Pressman, *Hill Street Blues* prefers not to compromise the quality of the script in order to meet the normal shooting schedule, which begins with a 7:00 a.m. call, and extends to a 7:00 p.m. wrap, or longer if the day's shoot runs into overtime — *Editor.*

POST-PRODUCTION SCHEDULE FOR "HILL STREET BLUES"

FIVE WEEKS BEFORE AIR DATE:

PRODUCTION PHOTOGRAPHY FINISHED: FIRST CUT BY PICTURE EDITOR (Six Days)	DIRECTOR'S CUT (Four Days)	PRODUCER'S CUT (Three to Five Days)

← PICTURE EDITING →

TWO WEEKS BEFORE AIR DATE: ONE WEEK BEFORE AUDIO DUB:

FINAL CUT — SOUND EDITING BY HORTA EDITORIAL (Five Days). CONCURRENTLY
MUSIC SPOTTING (One Day)
SUPERLOOPERS ADR SESSION (One Day)
FOLEY RECORDING (One Day)
MUSIC RECORDING AND EDITING (One Day)

← SOUND EDITING →

ONE WEEK BEFORE AIR DATE:

THURSDAY	FRIDAY	MONDAY	TUESDAY	WEDNESDAY	
RE-RECORDING AT GLEN GLENN FIRST (SILENT) ANSWER PRINT		SECOND ANSWER PRINT (WITH SOUND)	TRANSFER TO VIDEOTAPE CANADIAN AIR DATE	SHIP AIR PLUS BACK-UP COPIES TO NY	AIR DATE

← RE-RECORDING AND TELECINE →

Note: The length of each post-production stage is consistent from show to show. However, the gap between final dub and air date varies.

Associate producer Ellen Pressman (left) and co-executive producer Gregory Hoblit on the *Hill Street Blues* squad-room set.

Photography by Larry Blake

Hill Street Blues

range "presence" boost.) "The flat 816 is much more unforgiving than the standard 816," he says. "You can't get away with much, and your boom man must be good — he has to be right there on the money."

On most moving shots through the *Hill Street* squad room, Marky would employ plant microphones hidden throughout the set. "Very often I would have six to 12 mikes, and dance them across the panel, with my cable man repatching as we went." (Marky uses a four-input Sela 2880-BT mixer with his Nagra 4.2 recorder.) Favorite plant mikes include Tram lavaliers, Sennheiser MKH415 and -416 models.

During these squad-room scenes, and all others, the background extras remain quiet while production dialog is being recorded, often pretending to hold conversations. (If you think you hear conversations in the background, you're right; see the section below on the Superloopers.)

Although production mixers have been known to ask that actors and extras put "booties" over their shoes to cut down unwanted noise, Marky never liked to do this when he handled production mixing for *Hill Street* because it literally affected the actors' performances. "If you booty [actors] Dan Travanti, Kiel Martin, Taurean Blacque or Veronica Hamel, believe it or not, you are taking away from their performance. When Veronica would walk around the court room, she would go 'click, click' with her shoes as emphasis to her dialogue. Try to Foley that later!"

Associate producer Ellen Pressman notes that most extras wear soft-soled shoes anyway, and that "the real problem is paper and clothing rustle. When you have 25 people in the room, all that noise adds up. And when we get to the dub stage and it gets brightened, it sometimes sounds like you have a fireplace going on in the background."

Supervising producer Scott Brazil is intrigued at the possibility of using a stereo Nagra to record production tracks, utilizing the second track to capture synchronous minus-dialog footsteps and general background noises. He had used a similar technique while working on *The White Shadow*, whose basketball games were sometimes recorded with the coach on one track, and the basketball dribbles and shoes squeaks on the other. For *Hill Street*, Brazil thinks such a technique would be especially useful during roll call, with the "blues" and the desk sergeant recorded on separate tracks.

Supervising producer Scott Brazil

Picture Editing

After shooting ends, picture editor Joe Ann Fogle will spend five to six days assembling an episode into a rough cut for the director to see. (Fogle alternates shows with two other editors, David Saxon and David Solomon.) After this, in accordance with the latest Directors Guild contract, the director has four days to do his or her "director's cut," whose duration has to be within a minute of the final air time. Then the show is in the producers' hands, during which time executive producers Steven Bochco and Gregory Hoblit will map out and re-arrange the order of scenes on three-by-five cards. Everyone connected with the show agrees that *Hill Street Blues* seldom airs in the same sequence as it was written. "We change the acts in, acts out . . . *everything*," says Fogel. (Which is easy to understand, since the show has 14 main characters and three to six concurrent stories per episode.)

Production sound engineer Sunny Meyer

For the 1984/85 season there will be no more teaser/reprises ("Previously on *Hill Street Blues*") or teaser/trailers ("Tonight on *Hill Street Blues*"). Therefore, the program will be two minutes longer than last season, each episode now lasting exactly 49:19, to which is added approximately 10 minutes of commercials. Since there is 2:43 of "format footage" — opening and closing credits, and the "bumper cards" leading in and out of the commercials — an episode has only 46:36 (±0 frames!) of actual program footage.

The only constraint placed on Fogle regarding how the 60 minutes are divided up is that there cannot be a commercial break on the half-hour, to hopefully prevent channel switching in the middle of a show. Should an episode come in under the required length during the first cut (which rarely happens), a new scene or two will have to be shot and written *post haste*.

Fogle cuts *Hill Street Blues* on a standard upright Moviola for the same reasons that the "glorified sewing machine" is standard equipment for sound editing: "The accessibility to tracks is much greater because a Moviola is used in conjunction with a sync block with two sound heads. In addition, the ability to *move* the tracks is much easier than on a video editing system, or even on a KEM. It's simpler for me, but then I learned on a Moviola."

Sound Editing: Time Pressures

Sound editing for *Hill Street* has been handled by Horta Editorial, of Burbank, since the show's inception. Sam Horta oversees all the shows that his company handles, including *Cagney and Lacey*, *Remington Steele*, and *St. Elsewhere*. *Hill Street* is supervised by Gary Krivacek, with Eileen Horta handling dialog and ADR, and Dave West cutting sound effects.

Probably no other group more acutely feels the time crunch of television schedule than sound editors, who have very little time to do a lot of painstaking work. "We try to give them five days," says Pressman. "Sometimes it becomes two days and a weekend. And five days isn't even enough. Just cutting roll call and the Superlooper voice-over ADR [see below] is a *lot* of work. We try not to change the picture after we give them their black and white dupes.

"In a few rare instances we have had a first cut on a Friday, a final cut on a Sunday, and had to cut, mix and transfer the show to tape before the Thursday dubbing mix." (In their contract with MTM Productions, the cutters at Horta Editorial have to get their black and white dupes five working days before the dub, or else there are overtime charges.)

In May 1983, to aid in the sound effects editing of television shows, Horta Editorial purchased an Otari MTR-90 Series II 24-track, an MTR-10 two-track, and an Audio Kinetics 3.10 Q.Lock synchronizer. West uses the multitrack premix and assembly set-up to supplement the sound effects, Foley, dialog, and ADR that are cut on 35mm mag film.

"Because I can do some combining and premixing," he explains, "I will often use less tracks than you might have in 35. For example, instead of having a car in, idle and stop, spread across three channels, I will pre-dub and combine them to one channel. It saves both the number of units [35mm sound-element reels] that go to the stage, and dub time."

West mainly uses NAB cartridges for playing in non-sync background effects, to "back up whatever we are doing in terms of the volume of the sound." Whenever sync is critical, however, he goes to a quarter-inch library that includes copies of all of the cart background effects, in addition to specific sounds. On an average show, he will fill up a maximum of 16 tracks on the Otari.

Sam Horta thought he was going to encounter more problems with the use of the multitrack process during editing. "I am most happy that what we deliver to the dubbing stage is of higher quality," he concedes, "since we can EQ, clean up, and preview ahead of time. It's hard to imagine not doing it this way anymore, and I'm looking forward to updating the system and to being more creative with it. I think if we had seven days to do the show . . ."

There is some degree of overlap between what West lays onto his 24-track tape and what Supervising sound editor Gary Krivacek cuts on 35mm mag. As might be imagined, time constraints are an important factor during the sound editing process. Krivacek notes that "we try to give Dave the things that he can just pump out as quickly as he can. We will sit here and cut all of the hard [sync] effects, and scenes like chases and gunfights. Dave is excellent on those scenes, too, but sometimes he doesn't have the time." Some effects that are sync-critical and require meticulous matching, such as EKG beeps, are usually relegated to being cut on a mag on a Moviola.

A typical squad-room cut sheet will have three basic background tracks, with specific typewriters, phones, off-stage jail door open and closes, etc. added as separate elements. In addition, West has carts of dispatcher's voices recorded in New York, L.A. and Chicago. Whenever Krivacek has to cut squad room BGs, he often uses a 35mm three-stripe background that contains squad room "walla," typing, and footsteps on different channels. Access to this three-stripe is particularly convenient if dubbers are in short supply.

For Eileen Horta, the almost exclusive use of boom microphones on *Hill Street* saves time, since she doesn't have to remove clothing noises and pops that are endemic when using hidden lavalier mikes. She usually cuts production dialog for reel #1, the roll call, and also the Superlooper voiceover ADR tracks, a task which takes her a full day.

Four other editors take one reel of production dialog each, splitting the editor's "A-Master" onto three and sometimes four tracks in preparation for dubbing. As noted earlier, there is very little looping of actors' lines, although lines are often added.

Dave West premixing and assembling sound effects onto an Otari MTR-90 24-track at Horta Editorial.

Picture editor Joe Ann Fogle

Foley and Superlooper ADR are recorded on four-track 35mm, and then layed off onto separate rolls of 35mm stripe for fine-tuning sync. Foley for all five reels is "walked" in a day. (In movie-industry parlance, "walking" includes performing all Foley effects.)

Superloopers: "Background Voice-overs"

Once an episode has begun its week of sound editing, Pressman will spend a day doing "background voice-overs" with a band of actors known as *Superloopers*, some of whom regularly perform with the improv group "Off the Wall" at the Improvisation in Los Angeles. Superloopers has handled similar work on *Raiders of the Lost Ark* and *The Hand*, in addition to *The Yellow Rose* and *The Thorn Birds* for TV. According to leader DeVera "Dee" Marcus, the group originally was hired by Hoblit to "personalize the backgrounds of *Hill Street Blues*, and to augment the tempo and richness that you see on the screen; to make it specific."

The group's work includes duplicating the background sound of people answering phones in the squad room, plus providing small conversations during street and bar scenes. "We put voice-overs in everything, basically," says Pressman.

During the eight hours Superloopers spend at the Glen Glenn ADR stage, the group covers every *Hill Street* scene. This task often *includes* the scenes involving Frank Furillo and Joyce Davenport preparing for bed, for which Superloopers will often add the sound of radio or TV announcers in the background. Similarly, they often have to provide voices of TV game-show hosts, contestants and the like for scenes in which the officers might be patrolling tenement hallways, probably in search of a family disturbance.

Usually, two tracks are recorded from the Superloopers' session: a general walla, and another with more specific actions. Pressman likes to add specific backgrounds when she sees "something that would make a nice transition if we are moving across the squad room. I like to go for more specific things than just having a general walla track."

Roll call, which is the toughest scene in any given show to edit and mix, will sometimes have four or more Superlooper voice-over tracks. Often, background dialog is placed in a scene to add color. In many of these instances, the Superlooper tracks will help add verbal motivation to a character's actions that wasn't clear in the production track, and which would be far more effective than a library walla.

An example of this occurred when Lieutenant Henry Goldblume, who has retained Sixties idealism in face of the reality of police life, draws his gun on a group of gang members that was harrassing him while he fixed a flat tire in their neighborhood. According to Marcus, "This was a strong story point: he had been driven to draw his gun against his conscience. When it was filmed, they had excellent actors playing the hoods, but the *sound* of it wasn't that threatening. [Superloopers member] Carl Lumbly listened to the production track and improvised the part of an off-screen, deranged, homicidal hoodlum. By the time he finished the whole scene, it was terrifying; it justified to the audience Henry drawing the gun."

Pressman relates a similar incident last season during the aftermath of a gang war. "On camera, you saw gang members being dragged off, and only two of them were being paid to say anything. We put in sounds of gang members threatening one another, being harrassed. It was really eerie."

Although sound-effects editor Dave West does have pre-recorded cart loops of dispatchers, specific plot-oriented sections are looped by Wendy Cutler, who is also heard on the opening credits ("We have a 911, armed robbery in progress, C-Surplus store corner of People's Drive and 124th Street.") Another well-known voice on *Hill Street Blues* is that of Mick Belker's Ma, which previously was played by Dee Marcus. Recently, however, the producers have decided to leave Ma's side of conversation to the viewers' imagination, letting Bruce Weitz' acting carry the scene.

From left to right: Horta Editorial sound editors Sam Horta, Gary Krivacek and Dave West.

Music Recording and Editing

The title music for *Hill Street* was written by Mike Post, who also composes the underscore for every episode. When the picture cut has been locked, Post, Allan Rosen, the music editor, Gregory Hoblit, Scott Brazil, and Ellen Pressman screen the film, "spotting" where music should be added. After the screening, Rosen translates the spotting notes into specific timings to guide Post while writing his score.

"Usually on *Hill Street* we have a pretty tight schedule," says Rosen who, like the Hortas, has been with the show since the pilot. "We try to spot a week before the dub; they can cut it this close with *Hill Street* because there isn't an abundance of music."

Rosen also provides the staff at Horta Editorial with a copy of his spotting notes "just so they will know basically where music is going. But if there is a car chase, there's not much that we can do. It's going to be noisy, and just a matter of mixing the elements right. We're lucky to have a group of mixers who have a really good feel for the product; Ken Polk, the music mixer, is excellent.

"Many TV shows shoot with music in mind — a lot of long car drives, things like that. Some shows start the music at the beginning of the show, and go right through to the end. *Hill Street* cuts from one scene to the next almost instantly, so there are no long pauses. The music they *do* use is to emphasize a specific thing; it's *not* filler."

Scoring is done in Hollywood at either Group IV or United Western. A standard film three-track mix — strings on one track, percussion and rhythm on a second, and piano or synthesizer on a third — is recorded on both 35mm mag and half-inch tape. Rosen edits the music on mag, the half-inch being saved as emergency backup. Mike Post's original title and end-credit music was recorded 24-track and remixed to a master three-track.

Most scoring sessions involve between 20 and 25 musicians, including strings, guitars, percussion, piano, and synthesizers; brass and woodwinds are rarely used. Usually a single, three-hour session is sufficient to record the underscore music for an episode. As the show's post-production schedule would have it, *Hill Street Blues* is usually scored the day before the dub — one week before its air date.

Rarely is music outside of standard underscore used in *Hill Street*, although in a few episodes last season the writers had bagpipe and accordian players hired to flush rats from the Hill Street Station. The instruments were played on the squad room set by professional musicians, and sync-pulsed playback tapes made for the actors to follow.

In a memorable episode during which Howard Hunter tried to kill himself, great care was taken in choosing *the* source music that the lovable, erudite EAT (Emergency Action Team) leader would play in what he thought would be his last moments on Earth. Pressman and then associate producer David Latt spent a few days at the UCLA library listening to a selection of records before finally settling on Vivaldi's Sonata #4 in B minor, opus #5. The selection was orchestrated and conducted by Mike Post.

Final Dubbing Stage

Re-recording for *Hill Street Blues* takes place at Glen Glenn Stage R on the CBS Studio Center lot, Hollywood, where the series is also shot. Lead mixer Bill Nicholson, who handles dialog, has been with the show for the past three seasons; music mixer Ken Polk and sound-effects mixer John Asman joined the show two years ago.

It takes approximately 12 to 16 hours (a day and a half to two days) to dub each 49-minute episode. Although this time frame is miniscule by featurefilm standards, it is about twice as long as the average mix for the majority of one-hour TV shows. "On many shows, if they are not finished by four o'clock, they think something has gone wrong," says Polk. "A normal reason for taking more time might be pre-dubbing, but on *Hill Street* we are just sitting there fine-tuning everything."

Occasionally, extra time must be spent pre-dubbing because the Quad-Eight board at Stage R contains only

Superloopers performing "back ground voice-overs" at the Glen Glenn ADR stage.

36 inputs. Says Polk: "We spend a lot of time finessing very small details, like chair squeaks. Someone in the background will pick up a styrofoam cup and you have to hear it. It's getting more and more like a feature."

Nicholson notes that this concern for detail is true of all of the MTM shows on which the lead mixer works. "I think the MTM shows spend a lot of time in dubbing, and *Hill Street* set the example. They don't like to rush the dubs, and will spend time where other companies might not. They like to have that real sound and cover everything."

According to Polk, "One thing that's different with *Hill Street*, compared to other action shows, is that most of them tend to stress one thing during action sequences: music high and sound effects low, or vice-versa. *Hill Street* tries to make *everything* in an action sequence register equally well. And they don't dig any holes in there where you can hide the dialog. It just squeaks through so that the energy level is maintained.

"I wind up using a lot of limiters but, if we are given enough time to play with it all, it usually works. And they do try to take time on those sequences to make a blend; they want to be able to hear everything."

In Glen Glenn's machine room there are 20 Magna-Tech Electronics and four vintage Ampex 35mm playback dubbers, and a four-track MTE 35mm recorder. An Otari MTR-90 Series II is used to play back the 24-track

. . . continued overleaf —

Hill Street Blues

Re-recording at Glen Glenn Stage R, with (from left) John Asman, Ken Polk and Bill Nicholson handling effects, music and dialog.

sound-effects tapes built at Horta Editorial by Dave West, with sync lock maintained by running a reel of 35mm mag with pre-recorded timecode in sprocket interlock with the picture. In the event of picture cuts, offsetting the 24-track tape is as simple as resynchronizing the timecode reel. Thus, the mixers only use the standard film-chain motion controls; they are only aware of the presence of a multitrack tape in the machine room by a light that tells them when the BTX Shadow synchronizer has locked the Otari to the timecode track.

With pre-dubbing held to a minimum, all EQ, cleaning up of production tracks, in addition to combining a multitude of sound effects, must be done directly to the four-track master mix. Because they have four tracks at their disposal, the mixers split the sound effects into Foley, and everything but Foley. Thus, the master 35mm four-track contains dialog, music, and two effects tracks. (Is this a DME*E*?)

The crew at Stage R monitors through a standard Altec-Lansing Voice of the Theater speaker, a particular mix being checked through a small speaker only if there is a question of how music or loud sound effects might affect dialog intelligibility.

"They go for the big 'theater' sound," Polk says. Lead mixer Nicholson offers that one of the few differences between feature and TV dialog mixing is that when mixing for the big screen, "you can get by with more subtle [low-level] dialog."

Hill Street Blues is mixed to the "Academy Curve," which has been used for mono theatrical films for almost 50 years. Originally, the Academy curve was designed to codify the way in which Hollywood studios would combat the high-frequency optical noise revealed by the "modern" speakers of the mid-Thirties: the response is down 18 dB at 8 kHz. (By way of contrast, Dolby Stereo two-track, four-channel prints have a wide frequency response — to over 13 kHz — because of the A-type Dolby noise reduction on the print.)

Hill Street dubs with an Academy filter inserted in the monitor system because each show is eventually transferred to an optical soundtrack. Therefore, even though the mag master of *Hill Street* is used during the transfer to videotape for broadcast, the fact that an optical soundtrack is needed for distribution means that Academy monitoring is essential. However, NBC and all the major networks broadcasts to a wider than Academy response, the mag master is Academy filtered during transfer to videotape. Because of the mid- to high-frequency pre-emphasis that is applied during the mix to compensate for the Academy curve, if the mag master were not filtered during the transfer, notes Polk, "some of the stuff would crack your glasses."

Sound effects mixer John Asman feels that the technique of compiling sound effects on 24-track has helped him during the re-recording process. "Working with the 24-track has made my job a bit easier," he considers, "because you have 'all of the units built.' Dave West lays things out in a straight line; all of the backgrounds end up in the same channel throughout the whole reel. Whereas on 35mm, editors have a tendency to put the backgrounds in one section, the cars in another, etc; things are spread around more and you have to chase them. Another thing that makes my job easier is that the editors for *Hill Street* have been there since the beginning, and are well prepared."

Transfer to Videotape

While the sound is in its final stage of editing and mixing, the camera negative has been conformed to the editor's workprint, and a first trial print made, incorporating the desired color corrections. Technically, this is not a "first answer print," since it does not include an optical soundtrack.

Once the dubbing is finished, an optical negative will be shot from the single-track combine of the four-track master mix. Then, a second trial print will be made to include the soundtrack.

On Tuesday night before the Thursday airtime, Ellen Pressman will take the mag master and second trial print to Vidtronics in Hollywood, and supervises the transfer to a one-inch, Type-B videotape master. (Type-B is generally considered to offer better picture quality for mastering than the Type-C format utilized by the networks.) The mag master, and not the optical track, is always used. During this transfer, blank spaces are "slugged out" for commercials, so that the master videotape keeps running in real time during transmission.

A Rank Cintel flying-spot scanner is used to transfer the second answer print to videotape, and some might wonder why the producers do not utilize the cut negative itself, given the superior sharpness of negative-to-tape transfer. The answer harks back to the concept of "noise" desired by Greg Hoblit. In fact, tests indicated that a standard projection-style telecine gave an even "grittier" image, and would be used were it not for the precise, frame-accurate color correction offered by the Rank Cintel.

Four one-inch Type-C videotape dupes are then made from the Type-B master: an air and a backup copy for both New York and Los Angeles, where the Pacific Time Zone sees the program three hours later. Air and backup copies are played simultaneously to allow instant switchover should a machine go down while 20 million people are watching *Hill Street*.

Network television shows in Canada either air on the same day as the States or before, but never after, since the majority of the population in Canada is on the border and can pick up U.S. stations. After all, high-paying advertisers don't want to spend money on what would otherwise be the second screening. *Hill Street Blues* airs in Canada on Tuesday before the Thursday U.S. broadcast. The one-inch dupes for Canada have been shipped as late as Monday, but an effort is made to get them there by Saturday. Because of time factors, the Canadian one-inch master often has to be made from the first trial print.

Within the span of less than two weeks — the time many feature films spend on just Foley recording — the sound for an episode of *Hill Street Blues* is created: ADR and Foley recording, sound editing, music scoring and editing, and final mixing. While this tight schedule might be par for the course on episodic television series, the resultant sound quality is never compromised. Indeed, *Hill Street Blues* is one of the few shows which challenges the long-standing myth that film is simply a visual medium.

■■■

1. For a detailed look at this production philosophy, the interested reader is referred to the chapter on *Hill Street Blues* contained in *Inside Prime Time*, written by Todd Gitlan and published by Pantheon.

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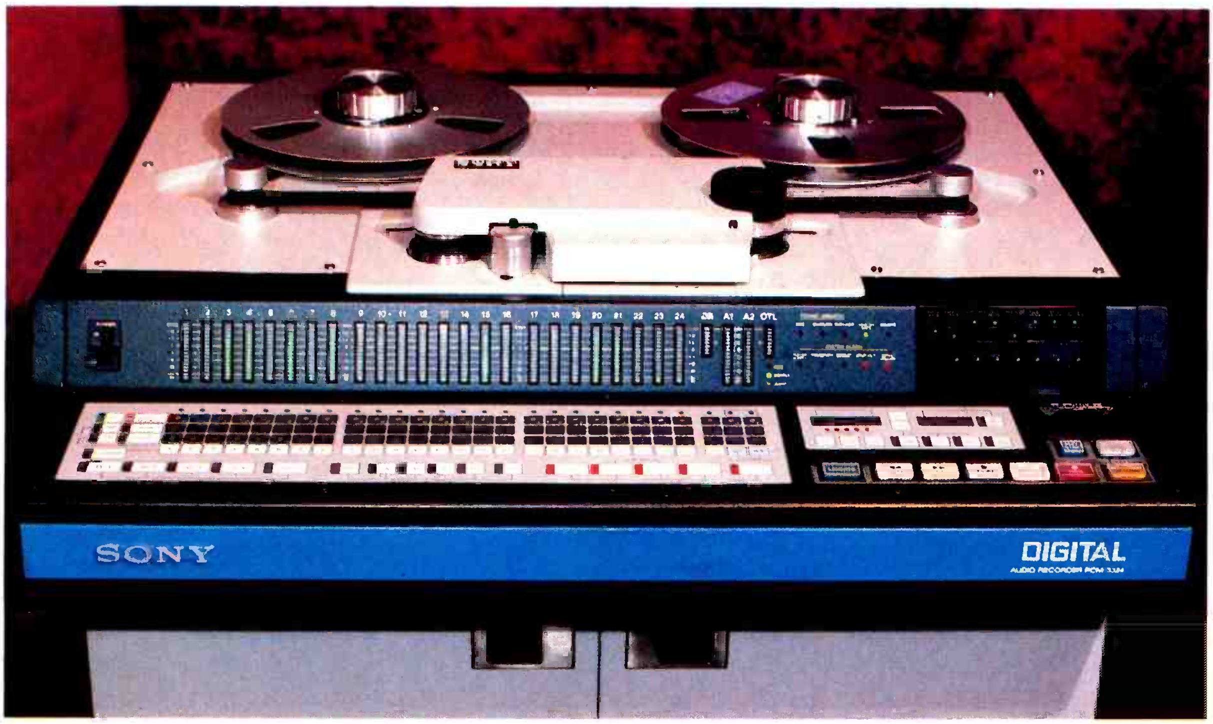

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STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

Northeast:

☐ **THE POWER STATION** (New York City) has upgraded with a new Neve Model 8088 customized console, which replaces the facility's seven-year-old Model 8068 in Studio A. According to chief engineer **Ed Evans**, "Our clients have been very happy with the way the Neve performs, plus we wanted to stay with that warm, Neve sound." Working closely with Evans, Neve designers installed customized features, including a cue arrangement that enables headphone mixes to be set up in less time than normally required. While Studio A's 8088 in-line console features 16 busses and 40 inputs, Neve has customized the desk so that the mix buss is still accessible even if a direct button is pushed on a given input. "This enables 24 to 48 track recording with little or no patching, even on the large sessions often booked in Studio A," Evans explains. Continuity of sound was considered an important consideration for Power Station management, says **Barry Bongiovi**, Power Station's technical director. "There was no question that we would replace a Neve with a Neve. You can't argue with client acceptance, and all the clients that have recorded in A so far have been very happy. We really purchased the Neve to please our clients and the engineers on staff." Other equipment in the remodelled Control Room A includes two Studer A800 multitracks, an Audio Kinetics Q.Lock synchronizer, a Studer A80VU half-inch machine with replaceable headblocks for two- and four-track mastering, plus several racks of outboard equipment, and Big Red monitors. The recent delivery of a new A800 24-track brings the total of A800s in use at the facility to six. According to Bongiovi, the newest A800 will be used part-time in Studio A, and part-time as a "floater" among the three recording and mixing rooms. The vintage Model 8068 console meanwhile has been placed in storage for another room the facility plans to build in the near future. *441 West 53rd Street, New York, NY 10019. (212) 246-2900.*

☐ **FAR AND AWAY STUDIOS** (Chester, New York) has added a Fostex B-16 16-track on half-inch multitrack, an Allen and Heath Model 1616D console, an MXR digital reverb, and Yamaha NS-10M "close-field" monitors. *Box 63, RD #1, Chester, NY 10918. (914) 294-7763.*

☐ **CRYSTAL CITY TAPE DUPLICATORS** (Huntington, New York) has added Dolby HX Professional modules to its Otari MTR-10 four-track master recorder. The modules automatically adjust the recorder's bias level, effectively extending the machine's high-frequency. *48 Stewart Avenue, Huntington, NY 11743. (516) 421-0222.*

☐ **ONOMATOPOEIA** (New York City) has added **Eric Eckstein** and **Ray Hopper** to its senior staff as engineer/producers. Eckstein, formerly with A&J Recording Studios, has a broad range of experience in commercial and industrial recordings, as well as in live recordings and documentary films. Hopper's expertise lies in the area of audio-for-video and audio-for-theater. He has had extensive experience with on-site audio installations with the National Shakespeare Company and the New York Theater Ensemble. According to **Matt Kaplowitz**, studio president, "The addition of Eric and Ray to our staff is a great asset. The diversity of their backgrounds will strengthen the range of services we can provide. Both men will be working closely with clients, designing and producing top quality soundtracks. And Ray's highly successful theater experience will give us a real edge in audio-for-theater." *37 West 57th Street, New York, NY 10019. (212) 688-3167.*

☐ **EASTERN ARTISTS RECORDING STUDIO** (East Orange, New Jersey) has added an AMS RM-X digital reverb, an MXR 01 digital reverb, a Yamaha DX-7 synthesizer, and a Roland MSQ-800 sequencer. *36 Meadow Street East Orange, NJ 07017. (201) 673-5680.*

☐ **AUTOMATED SOUND STUDIOS** (New York City) has completed a major upgrading program of its two-room facility, which specializes in jingle production. New equipment purchases include two Studer A800 24-tracks, a Dolby SP24 noise reduction rack, and four Studer TLS4000 modular SMPTE synchronizer units. The studio is also updating its current Studer A80VU multitracks with the newer "quick-punch" headblock assemblies. In addition to locking two multitracks together for 46-channel capability, Automated's new TLS4000 synchronizers will be used in conjunction with a JVC ¾-inch U-Matic video system for audio/video post-production applications. *1500 Broadway, New York, NY 10036. (212) 869-8520.*

☐ **BEARSVILLE STUDIOS** (Bearsville, New York) has announced that **Bill Robertson**, former chief maintenance engineer at Lucas/McFaul's Warehouse Studios in New York City, has taken over the position of chief maintenance engineer. *P.O. Box 135 Speare Road, Bearsville, NY 12409. (914) 679-8900.*

☐ **QUADRASONIC SOUND SYSTEMS** (New York City) has added to its in-house equipment with the purchase of a Yamaha DX-7 synthesizer, and a Garfield Electronics Doctor Click. In addition, the studio has added half-inch heads for its Studer A80 two-track. *723 7th Avenue, New York, NY (212) 730-1035.*

☐ **THE SOUND SHOP** (New York City) has completely redesigned Studio C, and installed a new custom Neve Model 5116 console. A departure from the facility's traditional approach of constructing its own audio consoles, according to studio president **Emil Neroda**, "With our 36-input Neve we are able to do something we always wanted — remix large-scale music productions down to stereo videotape or film." For videotape sweetening or dubbing stereo films, Neroda points out that the new console has unique monitoring features designed to handle the necessary punch-in demands. Vice president of operations, **Walter (Willy) Willumstad**, offers that one of the unique features is the provision of four stereo remix busses. "Usually the tracks in film mixing and frequently in videotape mixing are divided into three sections — dialog, music and effects," he explains. "In stereo work we need to work in stereo pairs — which would make six tracks for D, M and E — but we wanted even more capability to accommodate the advancing needs of Stereo Television and films. Neve built a 5116 console for us with preview [input] switching, soloing and other facilities for four stereo [eight mono] busses. We can mix a dual-language Stereo Television production or, in single language, provide for two sets of stereo special effects mixes, among other possibilities."


THE SOUND SHOP — redesigned Studio C

Tom Bush, the facility's coordinating engineer for the recent renovation, specified many of the console's features. "We have synchronizer-controlled automatic preview switching on any of the multitrack input/outputs," he says. "Neve were the only people who could make a console flexible enough to work in all three mediums: film, videotape or music mixing, plus conform with our unique style of working. In fact, three 24-track recorders and 36 [35mm] dubbers are tied into the console." The custom Model 5116 includes detented equalizers "for repeatability in videotape or film remix work," says Bush; and fully-variable equalizers "for flexibility in music mixing." Each input channel strip incorporates a limiter/compressor and a noise gate; plus pushbutton control of rerouting a subgroup output back into an input strip to enable signal processing of any part of a D, M or E mix. "Yet the console is so well laid out that an outside engineer can comfortably work according to his own style, either 'in-line' or traditional," Bush emphasizes. Remodelling of the new room has been under the direction of acoustic designer **Jim Falconer** — best known for construction of studios and remote truck at the Record Plant/New York — and who worked in close cooperation with Sound Shop's Emmy-award winning mixer-designer Emil Neroda, to ensure that the acoustics are still in tune with long-standing traditions in the film industry. Flush-mounted UREI Model 813 monitors powered by Bryston amps have been installed for monitoring. VCRs, multitrack recorders and 16/35mm projectors are located behind glass to diminish studio noise. The Sound Shop has a total of five studios, providing film and video mixing, ADR, Foley stage, and ACCESS digital effects editing. *304 East 44th Street, New York, NY 10017. (212) 573-6777 and 757-5837.*

☐ **SELECT SOUND RECORDING STUDIO** (Kenmore, New York) now boasts a remodelled studio and control room, designed by **Lakeside Associates**, and equipped with a fully automated MCI 636-Series console. Also added: a Lexicon digital reverb; Valley People Gain Brains; an Aphex Compellor; an MXR Pitch Transposer; ADA delay lines; Roland Chorus/Flanger; plus Audio+Design/Calrec SCAMP de-essers, dynamic noise filter gates, expander gates, automatic panner, and ADT module. Other studio equipment includes an MCI 24-track with Auto Locator III; an EMT 240; and an assortment of over 50 modern and vintage microphones. Studio equipment includes a Yamaha C-7 grand piano; Hammond B-3 organ; Pearce guitar amplifiers, Yamaha DX-7 synthesizer; Oberheim OBX synthesizer; and an assortment of Moog synthesizers. *2315 Elmwood Avenue, Kenmore, NY 14217. (716) 873-2717.*

☐ **CLINTON RECORDING** (New York City) has installed a Mitsubishi X-800 digital 32-track and an X-80A two-track machine; both decks were supplied by Digital Entertainment Corporation. Outfitted with the latest modifications, the X-800 is currently being used for underscoring duties for Francis Ford Coppola's upcoming film, *Cotton Club*. Engineer **Tom Jung** is working on the project at Clinton's Studio A, as well as some of his own recordings for his Compact Disc label, Digital Music Products. "With A&R Studios' large A-1 room gone," Jung explains, "Clinton's Studio A really fills a need for the larger production dates. And after being the first to use the original X-800 recorder back in 1982, I feel very comfortable in using both the equipment and the room." Studio president **Bruce Merley** also sees Clinton as filling a special niche in the jingle production business with digital recording. "More and more jingle producers are becoming hi-fi conscious," he offers. "Stereo Television and more sophisticated consumer television receivers will support the trend towards greatly improved audio-for-video." Clinton's recent digital upgrade represents the first X-800 digital multitrack available to the New York market. The next X-800 slated for delivery to the New York area is to be for Sid Zimet's **Audioforce** equipment rental company later this Fall. "We've been renting two Mitsubishi X-80 recorders to clients all over North America for the past three years with great success," Zimet considers, "and firmly believe that the addition of the X-800 to our equipment lineup will go a long way towards meeting our customers demand." *653 10th Avenue, New York, NY 10036. (213) 246-2444.*

CLINTON RECORDING — digital additions

Southeast:

☐ **MASTER SOUND STUDIOS** (Atlanta, Georgia) has installed a Solid State Logic 4000E console with the Primary Computer and Total Recall. According to owner/engineer **Bob Richardson**, "We are also doing some renovation in our present studios. We recently purchased the building adjoining ours to accommodate our high speed reel-to-reel and cassette duplication services, and are also planning to construct our third studio in the remaining space, hopefully by the end of this year or early in 1985." Featured equipment in the present studio facility includes hardware from MCI, BTX, Lexicon, dbx, Eventide, EMT, Orban, ADR, UREI, Poultec, and Dolby. *1227 Spring St., N.W., Atlanta, GA 30309. (404) 873-6425.*

☐ **LION & FOX RECORDING** (Washington, D.C.) recently added a Lexicon 224X digital reverb unit to the outboards in Studio A's control room. Chief engineer **Jim Fox** says the unit has proven to be invaluable in his 16mm film sound mixing and voice-over narrations, as well as its more expected musical applications. *1905 Fairview Avenue NE, Washington, DC 20002. (202) 832-7883.*

☐ **FLORIDA VIDCOM** (Pompano, Florida), a full-service audio/visual production company, has named **Mike Hoffman** as audio engineer and music director. A graduate of the State University of New York in music theory and composition, and Philadelphia's Combas College of Music, Hoffman is a jingle writer, prescorer, postscorer, instrumentalist on keyboard, bass and drums, and vocal talent. *3685 N. Federal Highway, Pompano, FL. (305) 391-4408.*

FLORIDA VIDCOM — Mike Hoffman

Midwest:

☐ **PEARL SOUND STUDIOS** (Detroit) has opened a new **Sierra**-designed studio that features Studer A80 MKIII 24-track and half-inch two-track machines, 40/24 custom Neotek Series II/III console with Valley People VCAs, a Lexicon 224X digital reverb, a 35 by 50 by 20 foot live room, and an assortment of vintage tube equipment. The studio is owned by independent producers **Ben Grosse** and **Geoff Michael**. *47360 Ford Road, Canton, MI 48187. (313) 455-7606.*

☐ **RED LABEL RECORDING STUDIO** (Chicago) has become the area's first facility to offer half-inch stereo mastering, following the purchase of a new Studer A80VU MKIII two-track. As part of a general upgrading program, the studio has also purchased a Quantec Room Simulator, and expanded the remix capabilities of its Harrison console to accommodate up to 44 channels, according to studio manager **Fred Breitberg**. *552 Lincoln, Winnetka, IL 60093. (312) 446-1893.*

☐ **CROSSTOWN RECORDING** (Kalamazoo, Michigan) has upgraded to 24-track capability with a new Otari MTR-90. Other equipment additions include a Lexicon Model 200 digital reverberator; Lexicon PCM-42 delay; Orban 422A limiter; Yamaha DX7 synthesizer; Oberheim drum machine; Simmons Clap Trap; and a three-camera CCTV monitoring system. *601 E. Crosstown Parkway, Kalamazoo, MI 49001. (616) 343-7972.*

PEARLSOUND — new studio opens

STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

South Central:

☐ **DALLAS SOUND LAB** (Irving, Texas) has named Canadian **Ron Cote**, staff engineer. While in Canada, Cote worked with producers Richie Cordell and Ken Kolokin, after which he moved to New York to accept a job with Kingdom Sound. It was at Kingdom that Cote worked as chief engineer on albums for **Blue Oyster Cult**, **Aldo Nova** and **Joan Jet**. He was nominated for the CARAS Awards — an honor equivalent to America's Grammy — and received several Gold and Platinum albums for his album work at Kingdom and Water St. Sound. *Three Dallas Communications Complex, 6311 N. O'Connor, Suite N-9, Irving, TX 65039-3510. (214) 869-7657.*

Mountain:

☐ **SOUND SUMMIT** (Lake Geneva, Wisconsin) is a new facility that offers 48-track analog recording with digital mixdown capability, using a Mitsubishi X-80A. The studio is equipped with two Studer A800 analog multitracks, plus A80 half-inch and quarter-inch recorders, linked to a Neve Model 8068 console equipped with NECAM automation. "We selected a Mitsubishi digital machine because of its growing reputation in the industry on both coasts," says **Phil Bonanno**, chief engineer and part-owner of Sound Summit. Summit will add the 32-track Mitsubishi X-800 if demand requires it, he explains. **Lakeside Associates**, which designed Lion Share and Fantasy Studios in California, has directed plant and acoustical construction at the facility, on the former site of Shadetree Recording. The new studio, which has been rebuilt from bare walls, has an improved, live sound, Bonanno says. *Highway 50, Lake Geneva, WI 53147. (414) 248-7666.*

Southern California:

☐ **FAST FORWARD RECORDING** (Hollywood) is a recently opened, independent 24-track recording studio aimed at album projects and jingle production. According to **Shepherd Ginzburg**, president and owner (*pictured right*), the studio features British-manufactured ACES equipment, including an ML-26/24 console — reportedly the only board of this type in the U.S. — ACES TR-24 two-inch multitrack, TR-2 half-inch and a vintage Ampex AG354 ¼-inch two-track. Monitor loudspeakers comprise a pair of JBL 8216s "mini-speakers," and a pair of Altec-Lansing 98228As. Ginzburg, who planned and constructed the studio from a shell, says the dimensions of his new facility measures 12 by 14 by 12 feet in the control room, along with two "mirror-image" rooms, measuring 17 by 11 feet each, intended for live recording of bands and drums, respectively. "We are a personable studio," Ginzburg adds. "I really don't want to compete with the bigger recording houses. I would like to open a number of smaller studios around the country, and record album projects for new acts." *6428 Selma Avenue, Hollywood, CA 90028. (213) 465-3457.*


FAST FORWARD — new 24-track facility

☐ **GROVER HELSLEY RECORDING** (Hollywood) is the new name of the Wally Heider-Filmways studio facility. The complex, which originally was the home of RCA Records' studio before becoming Filmways/Heider in 1977, includes two large studios (floor area 50 by 75 feet, with 30-foot ceilings) and a smaller (20- by 30-foot) room. Of the large rooms, Studio A is equipped for scoring to film or videotape, and Studio B for video-based scoring. Until its closure in 1982, the facility handled a full range of record sessions, film and TV scoring, plus assorted television specials and Music Videos. New owner, **Grover Helsley**, (pictured here with studio manager **Lisa Gerakos**) worked as a mixer in the facility during both the Heider and RCA days, and served as head mixer for the Filmways/Heider Scoring Services. "I had been interested in acquiring this studio for quite some time," Helsley comments. "We're going to do a little remodeling, but we're not going to change the acoustics of these great rooms at all. They were built years ago as multipurpose rooms with acoustics that are completely controllable; and that's how they will remain. We're looking to do every type of project; we have the space, the equipment and the personnel." Although the studio has been inoperative since 1982, the facility's original equipment was stored on the premises and will be used by the new operation. Control-room hardware includes three vintage, custom Neve consoles (one of which was partially designed by Helsley himself), five Ampex MM-1200 24-tracks, Sony video equipment, and three-track and mono 35mm film recorders. *1510 N. Ivar, Hollywood, CA 90028. (213) 469-6013.*


GROVER HELSLEY — old facility re-opens

☐ **LARRABEE SOUND** (Los Angeles) continues to update its equipment to complement the facility's Solid State Logic 56-channel, computerized console, including two Studer A800 multitracks, Mitsubishi X-80 digital two-track, an AMS digital delay, a Lexicon Super Prime Time DDL/effects processor, a Lexicon 224 digital reverb, two Valley People Kepex II noise gates, and dbx Model 160X compressors. *8811 Santa Monica Boulevard, Los Angeles, CA 90069. (213) 657-6750.*

☐ **COMPACT VIDEO SERVICES** (Burbank) has installed a new, custom-designed Neve 48-input series 5116 console in Studio A's control room. Equipped with NECAM 96 computer-based automation, the console is said to offer several sound post-production technical features not previously available, including a color video display with enhanced graphics; the ability to handle tape machine tach pulses, as well as SMPTE timecode data; group mutes and off-line group fader manipulation; and, 256k of on-board, random-access memory. Modifications to the console were developed by Compact's sound engineer, **Kelly Kotera**, in conjunction with Neve engineering staff. "Our advanced installation will now enable Compact Sound Services to generate an eight-stripe stereo mix with separate music, effects, dialog and audience for foreign distribution," states Kotera. "The new console will also allow us to do the finishing of film shows on videotape with the use of one mixer. This method of sound post-production may well become the most economical and efficient way to complete the track mixing on all film television shows." *2813 West Alameda Avenue, Burbank, CA 91505-4455. (818) 840-7000.*

☐ **THE RECORD PLANT** (Hollywood) has purchased a Sony PCM-3324 digital multitrack, which is slated for record projects, remote recording and film scoring. "The demand for the Sony multitrack is so heavy," states **Chris Stone**, studio president, "that we have had to rent additional machines in order to fulfill the requirements of our clients. Fortunately, a number of studios in town have made their PCM-3324s available to us. The time for digital audio has arrived." *8456 West Third Street, Hollywood, CA 90048. (213) 652-0248.*

. . . continued overleaf —

STUDIO UPDATE — FACILITIES, EQUIPMENT, PEOPLE

☐ **SUNSET SOUND** (Hollywood) has formally re-opened Studio One, following the recent installations of a custom-designed console with Neve NECAM II automation. (See April issue of *R-e/p* for complete details of the logic-controlled "super-board's" topology and operating parameters.) Other control-room hardware includes a pair of Ampex MM-1200 24-tracks; three ATR-102 mastering decks, including one with half-inch capability; JBL/Augspurger monitoring, including the new JBL Bi-radial HF tweeters; H+H V800 and V500 MOSFET power amplifiers and custom crossovers working with Klark Teknik DN30/30 equalization; and new outboards, including a Lexicon Prime Time II DDL/effects processor, Valley People Kepex II gates, AIWA Model 660 cassette decks, and an Audio Kinetics Q.Lock 3.10 SMPTE synchronizer with Genlock software option. Acoustic design for the new area was by **George Augspurger**, **Howard Weiss**, and Sunset's in-house staff. A Sony 25-inch Pro-feel video monitors provides playback for audio-for-video sweetening. Recent upgrades in the recording, which now features variable acoustics, include four, 64-input mike panels, each of which access the four independent stereo cue monitoring systems. According to session engineer **Humberto Gatica**, who is currently working in Studio One with Kenny Loggins on a new album project, tentatively titled *Fast Forward*, "Besides having everything that I want in a room — API Type 550A equalizers and NECAM automation — the new room is extremely comfortable to work in, and has a very clean and detailed monitoring system." *6650 Sunset Boulevard, Hollywood, CA 90028. (213) 469-1186.*


SUNSET SOUND — Studio One unveiling

☐ **LION SHARE RECORDING STUDIOS** (Los Angeles) has completed renovation of Studio B with the installation of a 48-input Neve Model 8128 console with NECAM II automation, and a new Studer A800 24-track. Compared to the Model 8108 installed in Studio A, the new Model 8128 features an improved solo system; enchanced monitoring; individual fader change-over switching; individual channel solo safe switching; and a comprehensive overdub facility for routing a mix of replay and mike signals to the monitor via a "sync" switch. The 8128's configuration is also said to allow engineers to use the tracking section of the console for effects mixes more effectively during mixdown. According to studio manager **Terry Williams**, after Studio B is in operation work will begin on a $3 million renovation of Studio C. While the planning of Studio C is still in the preliminary stages, serious thought, Williams says, is being given to Neve's DSP digital console which, linked to the facility's present Mitsubishi X-800 multitrack and X-80 mastering machine, would make Lion Share this country's first "all digital" studio. It's stil too early to say for sure," Williams cautions. "We love both our Neves and we plan to stick with Neve in the future." *8255 Beverly Blvd., Los Angeles, CA 91356 (213) 658-5990.*

Northern California:

☐ **ROBERT BERKE SOUND** (San Francisco) recently moved into its new, commercial sound production facility after 13 years at its previous location. With studio design by **Randy Sparks** of **Sonic Landscapes/Architectural Acoustics**, the two-studio production complex features acoustically identical control rooms. Although the two control rooms are dissimilar in size, both are said to exhibit a frequency response of ± 3dB (31 Hz to 16 kHz) without monitor EQ. In addition, excellent diffusion throughout the control rooms is said enable the "sweet spot" to be expended to include virtually the entire room. According to studio manager **Mark Escott**, "The new facility was designed to meet the specific needs of radio/TV, multi-image, and video producers; our emphasis on the final assembly of sound effects, music and voice elements is enhanced through the use of state-of-the-art equipment and computerized electronic editing." Equipment installation was supervised by **Mark Eckert** of **Sound Genesis**. Control Room A was primarily designed for radio/TV spot production and A/V soundtracks, and is equipped with an Auditronics 110-8 console, an Otari MX-5050 MKIII 8-track with dbx noise reduction; two Otari MTR-10 two-tracks; and an Otari MX-5050 ¼-inch four-track. Outboard equipment includes MasterRoom reverb; Symetrics phone-patch system; Orban compressors, sibilance controllers, filters and equalizers; MICMIX Dynafex noise reduction system; Bryston control room amplifiers; and JBL 4430 speakers. Control Room B was specificlaly designed for video soundtrack production, and is equipped with an Auditronics 700 Series console with VCA faders and six subgroups; an Otari MTR-90 16-track; two Otari MTR-10 two-tracks equipped with center-channel timecode; and Otari MTR-10 half-inch four-track; and an Otari MX-5050 four-track. *50 Mendell #11, San Francisco, CA 94124. (415) 285-8800.*


ROBERT BERKE SOUND — location change

☐ **BROKEN ARROW RANCH** (Redwood City), **Neil Young's** personal-use studio, recently purchased two Sony PCM-3324 digital 24-tracks and a RM-3310 synchronizer/controller. Comparison testing of available digital multitrack systems was conducted for Young by producer/engineer **Elliot Mazer**, a consultant on digital audio to the Center for Computer Research in Music and Acoustics at Stanford University. According to Mazer, acoustical quality, ease of operation and reliability were considered in the testing, which was confirmed through the use of the rented Sony equipment. "We'd been renting the PCM-3324 for almost two years and the experience has been great," he explained. 'I told Neil to go ahead and buy the Sony multitracks and get rid of his analog equipment. There's no reason why I would want to use an analog machine ever again. Also, Neil doesn't have a full-time maintenance man, [so] it was very important to have a machine that is reliable. I've already logged more than 300 hours on the Sony multitrack with just about every imaginable instrument. The sonic realism is truly extraordinary. The Sony multitrack has a very clear top, and the bottom-end goes much lower than the other multitrack we tested." *Redwood City, CA.*

People's Republic of China:

☐ **YUNNAN RADIO/TV SERVICES COMPANY** has ordered another Harrison MR-3 console through the Studer-Revox Far East Ltd. dealership. The recent sale brings to eight the total number of Harrison consoles installed at various broadcast and recording facilities within the People's Republic since 1978. The latest MR-3 console, which has a 36-position frame containing 36 input modules that route to 24 multitrack busses, is used for radio and television production and post production. Harrison consoles are currently being used in the PRC at Radio Guandong, Radio Guizhous, Radio Yunnan, Radio Heilongzhang, and Radio Shantung. The Pacific Audio and Video Company, the first organization in the PRC to order a Harrison console, utilizes a 32C Series board. Yunnan received its first MR-3 earlier this year, and is scheduled to receive three more by November.

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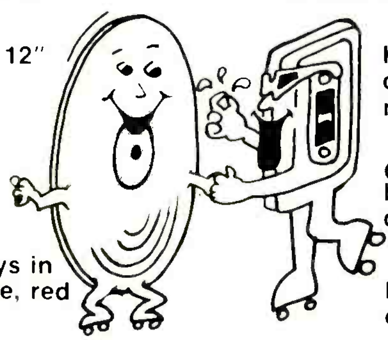

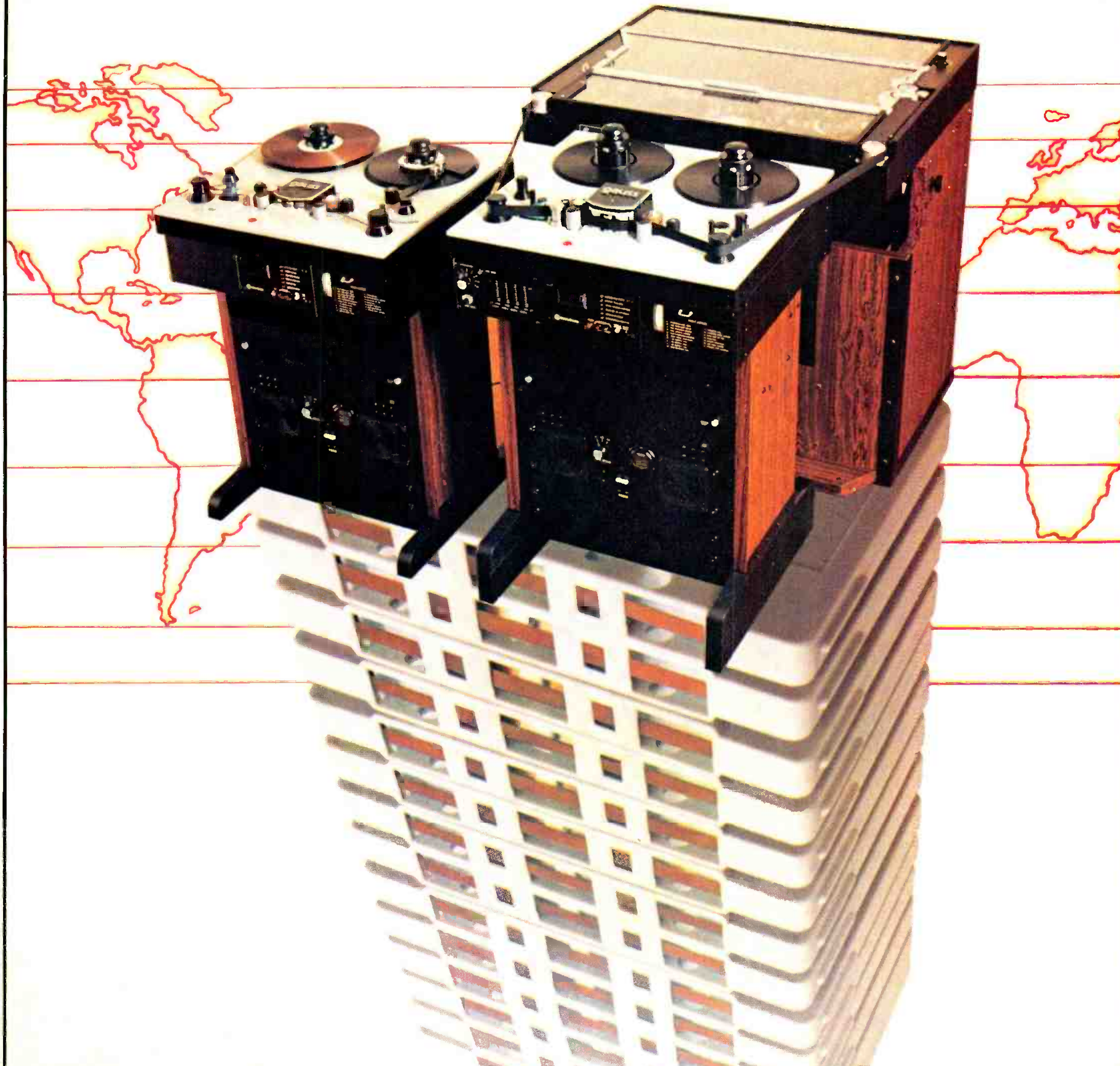

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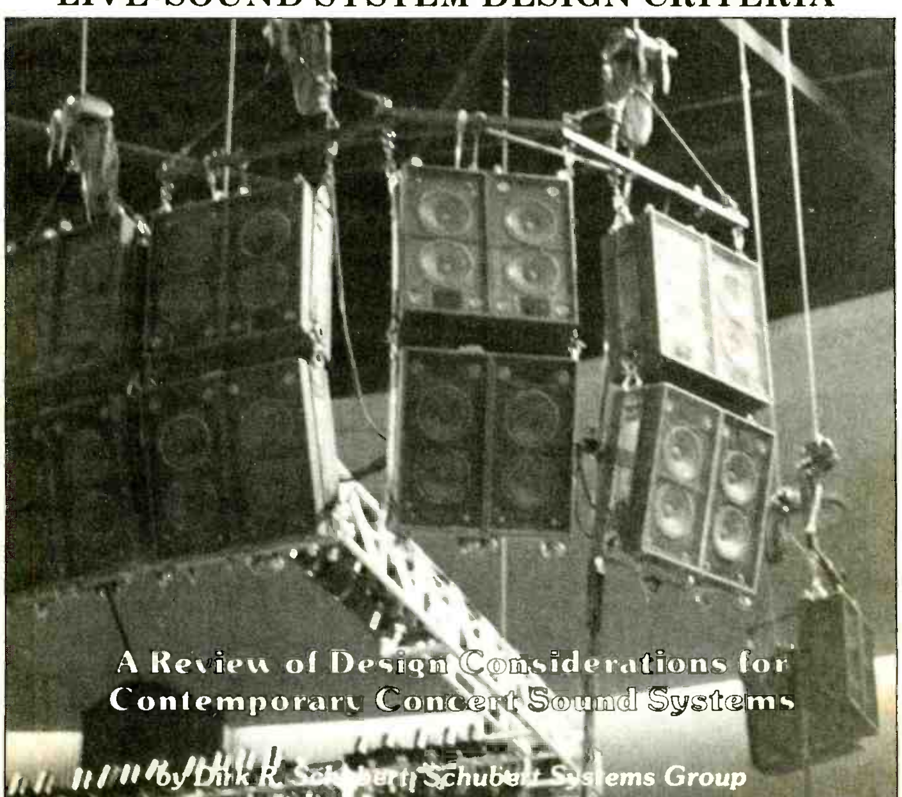
SCIENCE AND THE PERFORMING ARTS

LIVE-SOUND SYSTEM DESIGN CRITERIA



In the Seventies, when the music industry found itself in a fast growth period, all that a band or a band's engineer could hope for in the way of a PA system was one that worked. If it worked every day, that was even better.

The Eighties have brought new light into the sound reinforcement industry, by way of dramatically improved mixing consoles, digital effects, and high-speed amplifiers. Musicians and engineers are appreciating quality audio in their homes and studios, and are demanding to bring high quality sound reinforcement on the road to their audiences.

Today the buyer has a wider choice of available systems; making that choice is the subject of much controversy. A listening audition is commonplace, but a background of information regarding the physics of transducers and arrays can also aid in making a valid decision. This article will review the various design criteria of concert-sound systems, and attempt to show how science is being applied to the art of live-sound equipment selection. The wide variety of concert systems available makes it more important than ever for the buyer to understand the physical principles involved in system design. The design concepts presented here place particular emphasis on crossover design, direct-radiating low-frequency enclosures, and speaker array orientation.

Performance Characteristics of PA Systems

Technical performance criteria are used to evaluate three areas of the system (subsystems): crossover, amplifiers, and loudspeakers. A total of three check marks is available for each criterion; these checkmarks, listed in Table 1, show the subsystem's degree of significance within each group.

Re-arranging and totalling the table checks to show the significance of each subsystem yields:

Speaker system: five of six categories — eight check marks total.

Crossover system: four of six categories — six check marks total.

Amplifiers: three of six categories — four check marks total.

The performance characteristics of the system then are attributable to one or more subsystems of the PA; in fact, only one characteristic, system headroom, is traceable to all three subsystems.

In addition, we have designations for each frequency range in which we evaluate the above six performance parameters: bass or low-frequency; low-mids; midrange, and high-frequency.

System Dispersion and Directivity

Dispersion is the only characteristic of a PA system that is fully determined by one system — loudspeakers. A speaker system requires the organization of several different components to provide a near uniform dispersion pattern, so that the response does not vary greatly at any specific point in the room. The room itself is a large part of this problem. Not only can it be of various sizes and shapes and construction materials, but in many instances the PA must be flown at different heights to accommodate sight lines. In addition, the mixing position may not be optimally sited.

The system must sound the "same" at all locations; this is not necessarily equal sound pressure level at all locations. Generating a uniform dispersion pattern when trying to combine woofers (cone drivers), midrange horns, and tweeters, presents a real design problem. Few companies have really addressed this problem rigorously.

Dispersion, as a characteristic of a transducer, can be expressed as a vector: It has magnitude (SPL) and a direction (source). If we take a point source in free space, such as an isotropic radiator, to be our basic model of a perfect acoustic radiating surface, the only modification required to make it an ideal PA system is to limit the radiated space to that which only meets our coverage requirements. This is another function of the system design, since the actual space cannot, of course, be limited. What are the characteristics of a point source that make it a good model? The answer is: phase alignment, uniform dispersion, and integrated source. A PA system consists of many real sources (transducers) that must be configured in such a way that the characteristics of the point source are preserved.

A point source has wide dispersion because it is small in comparison with the wavelength emitted — a phenomenon known as diffraction. Diffraction provides wide dispersion, because the waves emanating from the small source have not been emitted in any

At Last, a 200 Watt Coax!

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Sound Ideas for Tomorrow... Today!

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particular direction (Figure 1). This concept can be approximated in two dimensions as a phased array (line source) perpendicular to the plane of desired dispersion (Figure 2). Phase alignment and integrated source can only be approximated due to the finite dimension of the "real" transducers.

Referring to Figure 3, which demonstrates the time error in the multiway source, the deviation of a real multiway source from a point source at the point of reference, R, is the distance S. The associated time-related error is given by:

$$T_{error} = [\sqrt{(S^2 + R^2)} - R]/V$$

where V = velocity of sound.

Taking a different view of transducer dispersion, it can be looked upon as an aperture or window. When the source wavelength is larger, the aperture radiation is more omnidirectional; when the source wavelength is smaller than the aperture, the radiation becomes more unidirectional. For cone drivers, the effective piston diameter can be thought of as the aperture; for horns, the actual rated dispersion angles can be considered the aperture. Due to diffraction, doubling the size of the aperture will result in approximately one-half of the dispersion angle, provided that the apertures are in the same plane. Along with the reduced dispersion comes a corresponding increase in directivity, as shown in Figure 4.

Figures 5 and 6 demonstrate that vertical arrays yield good horizontal dispersion, while horizontal arrays yield good vertical dispersion.

Directivity is the ability of a system to restrict dispersion to a particular area, and prevents the sound from folding back onto the stage (which causes feedback and can lower overall dynamics). Directivity also keeps PA sound from ceilings and other unwanted reverberant fields. High directivity may be achieved one of the following ways: 1) by assembling a number of horn-loaded or high-Q boxes, and aiming them at specific areas; or 2) by *arraying* a number of direct radiator or low-Q boxes. These approaches represent the two major design differences between systems. The stance of this article is that the latter — the use of an array of direct-radiator cabinets — more closely emulates a point source.

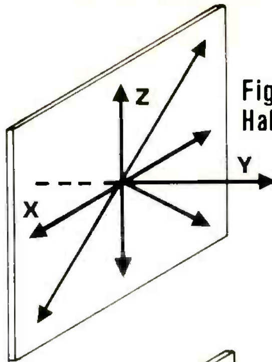


Figure 1: Point Source Radiating into Half-Space.

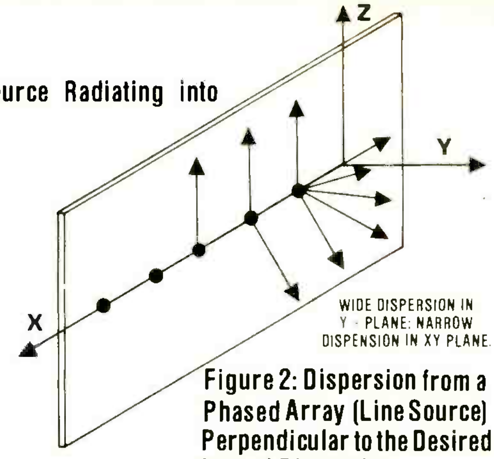


Figure 2: Dispersion from a Phased Array (Line Source) Perpendicular to the Desired lane of Dispersion.

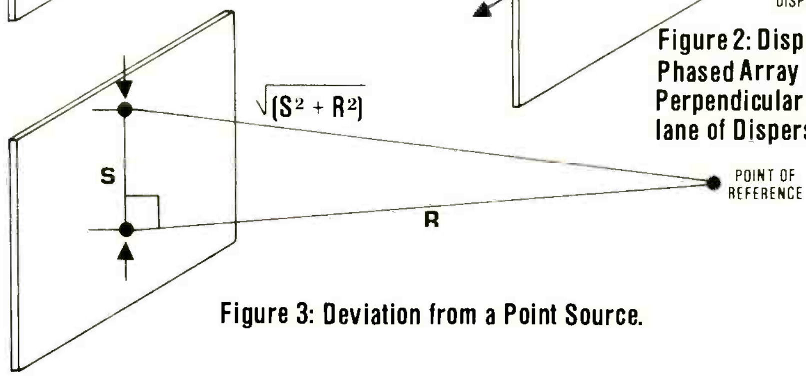


Figure 3: Deviation from a Point Source.

Low-Frequency Range

Reproduction of low frequencies (less than 125 Hz), being fully omnidirectional in nature, is simply the accurate movement of a large volume of air.

There are two types of low-frequency cabinets generally in use: horns and vented boxes. Actually, sealed-type cabinets make up a third type, but their efficiency is undesirably small for low-frequency sound reinforcement applications.

Horn-type cabinets attempt to increase efficiency by forcing the speaker to radiate into reduced space, in an attempt to increase SPL in a specific area. If we could take all of the output of our point source and direct it forward only (half space), we would see an SPL increase of 3 dB. If we could further reduce the radiation space to quarter space, we would see a further increase of 3 dB, and so on. When the wavelength of the source exceeds the length of the baffle, then the directing property of the baffle is lost and the dispersion is again omnidirectional.

Bass horns have several inherent problems, the first being limited bandwidth at low frequencies. When the wavelength of the source is greater than four times the length of the horn, zero power is transmitted by the horn. The effect of this is that the cabinet actually changes from horn-to sealed-type at a frequency within the passband. (This occurs at F = V/4L, where L is the length of the horn, and V is the velocity of sound.) Increasing the mouth area of multiple-horn systems by stacking does not change this transition frequency; it is a function of horn length only.

Another undesirable characteristic of bass horns is second-order harmonic distortion, although it should be noted that such distortion is not due to poor horn design or transducer quality. Instead, the distortion is a phenomenon that occurs due to the

TABLE 1: Technical Performance Criteria of the Three Subsystems of a PA Setup. (Note that where the particular performance characteristic is insignificant for a particular subsystem — in other words, currently available hardware and electronics have reached such a degree of perfection — it can be scored as zero.)

Performance Characteristics	Crossover	Amplifiers	Speakers
1. Distortion		**	*
2. Frequency response	*		*
3. Transient response	**	*	
4. Dispersion			***
5. System headroom	*	*	*
6. Phase alignment	**		*

— the Author —
Dirk R. Schubert, BSEE, with over 13 years experience in sound reinforcement, was a participating design engineer on the Gamble consoles, and currently serves as chief engineer for Schubert Systems Group, an R&D-intensive sound company providing service to a wide clientele of artists. SSG designs and manufactures its own speaker systems and specialized electronics exclusively for on-the-road applications.

high SPL in the horn's throat. The high SPL causes the air in the throat to become nonlinear, and actually rectify the signal to varying extent dependent on level. At low frequencies the distortion products generated can actually exceed the SPL of the fundamental, because they occur in a range in which the horn is operating at calculated efficiency, and the fundamental is below this range. The second-order harmonic distortion caused by non-linear air flow in the horn throat can be calculated from the following formula:

$$\%HD_{second} = [(K \times F)/(100 \times F_c)] \times \sqrt{I_t}$$

where K is a constant set by the horn geometry (i.e., exponential, conical, radial, etc.); F is the operating frequency, and I_t is the horn throat input power (watts per unit area), and F_c the cut-off frequency of the horn. This second-order distortion can be perceived as "false punch" or "edgy" sound.

Vented cabients radiate directly into half space until the baffle circumference is exceeded by the wavelength of the output. At this point the output does not decrease; the radiation simply returns to omnidirectional. When multiple cabinets are placed adjacently, the effective baffle circumference increases more slowly than the real baffle circumference because the source is not centered perfectly on the baffle. In addition, the larger cabinet volume of the vented enclosure keeps the woofers operating in a more linear (low distortion) mode.

A comparison of these two cabinet types shows that at lowest frequencies (below 60 Hz), vented cabinets are actually more efficient and show far less distortion than horn enclosures. In general, the low-frequency response of horns systems is cabinet determined, whereas the response of vented systems is determined by the response of the woofer itself.

The crossover also affects low-frequency response. More often this effect is seen through the phase relationship of the low-frequency crossover and the system's low-mid band. In certain high-order networks of Butterworth or Chebychev filters, transient induced ringing can occur in the crossover, which actually feeds incorrect signals to the amplifiers. This phenomena is not correctable by any cabinet design. Transient-induced slewing is also determined by the crossover, and affects the "tightness" or control of low frequencies.

Amplifier considerations for low-frequency applications include large amounts of RMS power, low output impedance, and high energy storage of the power supply (particularly under deficient line-voltage conditions).

Loudspeakers are generators as well as transducers, particularly at low frequencies, and generate a considerable amount of energy by their movement; this Back-EMF can be seen at the speaker terminals. The significance of this energy is determined by the source impedance of the amplifier and associated cabling: the lower the source impedance, the less significant the effect of Back-EMF. To avoid transmitting this energy into another speaker, separate low-impedance lines must exist from each speaker to the amplifier (Figures 7 and 8). The use of low-impedance lines is even more important with longer cabling associated with flying systems.

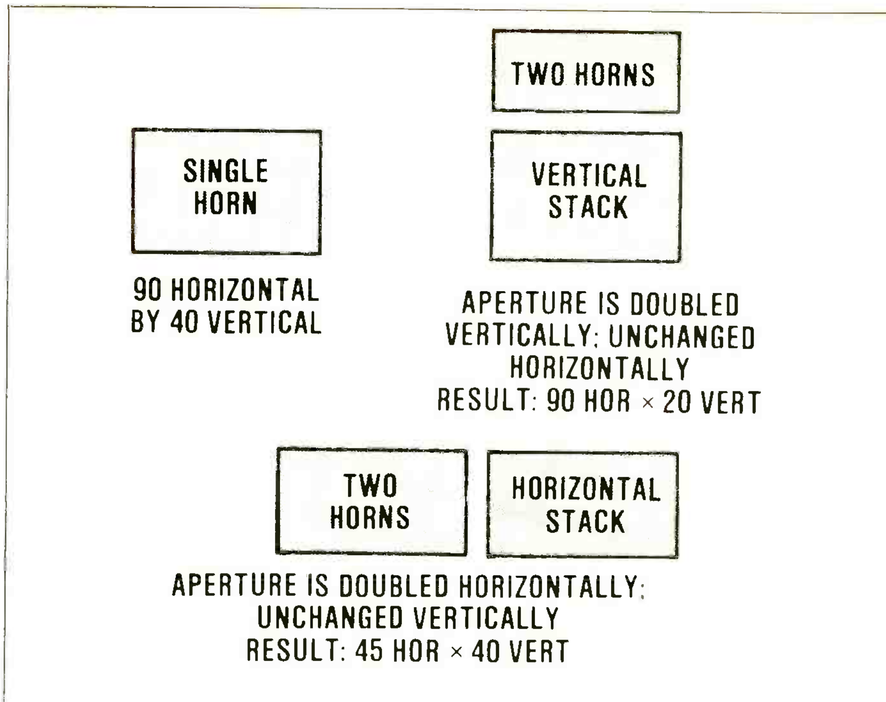
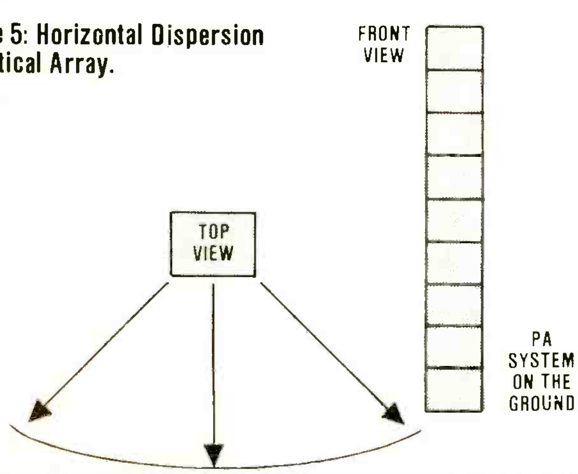


Figure 4: Effects of Vertical and Horizontal Horn Stacking on Dispersion Patterns.

Figure 5: Horizontal Dispersion of Vertical Array.



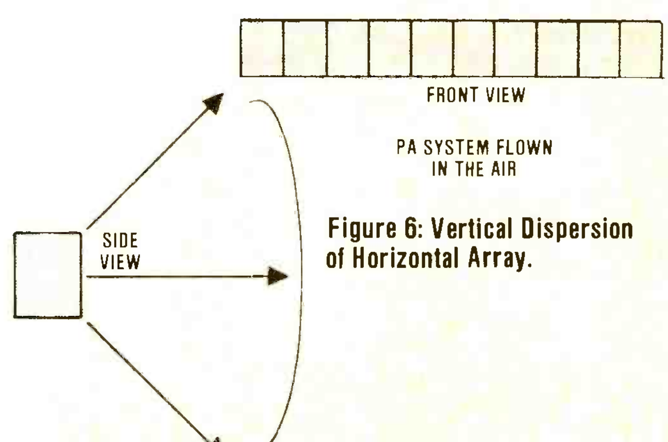


Figure 6: Vertical Dispersion of Horizontal Array.

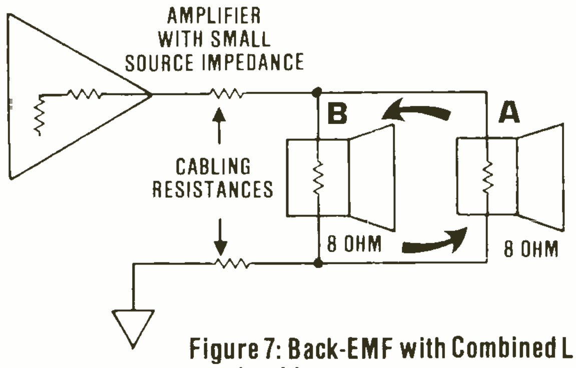


Figure 7: Back-EMF with Combined Loudspeaker Lines.

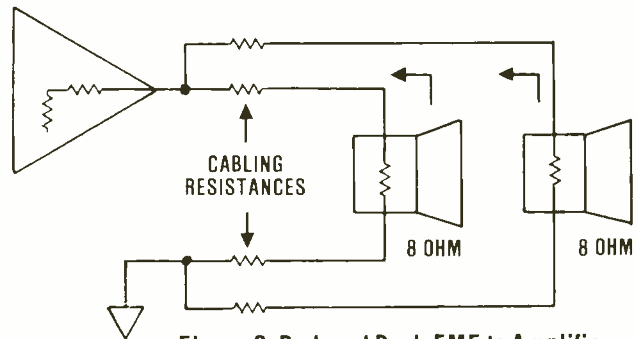


Figure 8: Reduced Back-EMF to Amplifier by Using Separate Loudspeaker Lines.

Low-Mid Frequency Range

The low-mid spectrum — between 125 Hz to 250 Hz and 800 Hz to 1.5 kHz — contains most of the program source's fundamental frequencies. *Reproduction of these frequencies, more than any other band of the system, shapes the characteristic "sound" of a particular PA system.*

Low-mids comprise most of the fundamental frequencies of all vocals and instruments, and many of the fundamentals and transients for drums. Band-edge response of the low-mids significantly affects the overall response of both the bass and the midrange. The lower range of this band has the excursion requirements of a woofer, and therefore such drivers are most often used. The presence of large SPL transients in the program material leads to many transducers being used to reproduce the lower range of the low-mid band without excessive distortion. The arrangement of this collection of transducers becomes part of the PA design problem.

A common problem with cone drivers is that they change dispersion through the band; their pattern narrows with increasing frequency. Horn-loading of the low-mids is feasible from a wavelength and mouth area standpoint, and can actually be beneficial in reducing the dispersion at lower frequencies to more approximate the reduced dispersion of the loudspeaker at higher frequencies. (This phenomenon happens naturally in a well-arrayed, multispeaker director radiator system.) However, the transient ringing and increased harmonic distortion of horn-loading can outweigh the controlled dispersion advantage.

With a large system the difference in acoustic output between a horn-loaded and a direct-radiator array is above and beyond the threshold of pain. The most linear operation of the transducer will occur in a highly damped, sealed cabinet. The linearity and accuracy of any low-mids system are most evident at the lower crossover point, where output of low and low-mid transducers must match if the full impact of the program is to be preserved.

Amplifier requirements for the low-mids are simple: the bigger the better. Amplifier clipping results in distortion products that are very audible through the speakers. In addition, the signal causing the amp to clip stands out in the mix, because the amplifier output has stopped tracing the input signal while clipping occurs. A misconception exists that a speaker rated for a given power cannot survive on a larger amplifier — this is *not* true. A speaker's power rating refers to the *average* RMS power handling capability, and does not take into consideration the dynamic range of the program material.

Midrange Frequencies

The midrange frequencies — running from between 800 Hz and 1.5 kHz to between 5 and 9 kHz — make up the upper fundamentals, and constitute the majority of the harmonics in the program. For the midrange area compression drivers are used by almost all companies, although there is a great variety of horn types and configurations that can be used to achieve adequate coverage.

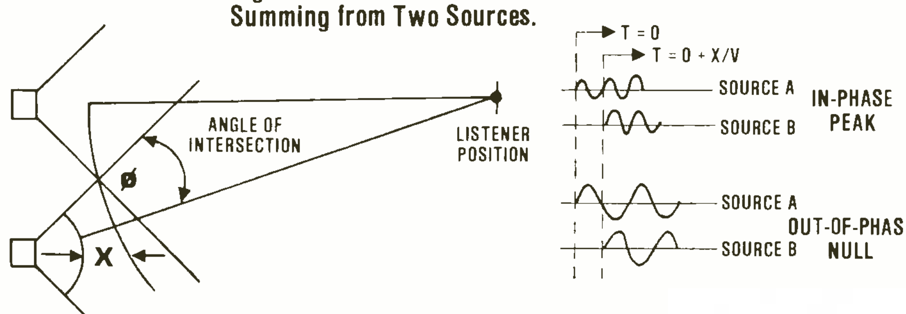
The problems encountered in achieving correct dispersion of midrange frequencies are related to the peaks and nulls caused by overlapping coverage patterns of horns. Peaks are increased SPL at certain frequencies due to acoustic summing of multiple sources in phase, while nulls are areas of decreased SPL caused by acoustic summing of multiple sources out of phase.

Consider two sources and a listener detailed in Figure 9. The distance differential, X, between the listener and the two sources will cause peaking to occur at frequencies with wavelengths equal to kX, where k is any positive integer; nulls will occur at frequencies with wavelengths of kX/2. These deviations in SPL occur at different frequencies, of course, depending on the distance, X. In general, the more pattern overlap (i.e., the sharper the angle of intersection of the patterns), the more pronounced the peaking. Increased numbers of horns with overlapping patterns will intensify the effect; to minimize this effect, horns may be arranged tangent to points of a cylinder, as shown in Figures 10, 11 and 12.

It is important to note, however, that these phenomena of acoustic summing are problems encountered mainly during outdoor performances, since for indoor performances the hall reverberation greatly masks the effect.

Horn specifications are typically given as 3 dB loss angles in the vertical and horizontal planes, and are quoted for a single horn. In a large system, to allow for a large enough number of horns for a given horizon-

Figure 9: In- and Out-of-Phase Acoustic Summing from Two Sources.



tal coverage without too much interference between units, the horns must have sufficiently small horizontal beam width. (The vertical beam width will remain that of the horns in the array.) Midrange diffraction, when used in conjunction with diffraction of the low-mids, equalizes (gain and frequency content) all points in the pattern through both ranges.

In a PA system, no negative feedback exists to correct or even detect distortion generated by the transducers. The accuracy or linearity of a transducer is directly proportional to its high-frequency response, or ability to "trace" the input signal. When a transducer is operated near its upper 3 dB limit, harmonic, transient, and other slew-induced distortion increase rapidly. There are several compression drivers available now that have response to 12 or 15 kHz; this is their mechanical 3 dB limit, where the diaphragm can no longer move fast enough.

In addition, there is also throat distortion caused by the same nonlinear air we encountered in our bass horns (compression driver throat SPL can reach 160 dB), which increases with frequency. This distortion, although predominately second-order harmonic, is very audible, and is perceived as harshness or "screech" in many three-way systems using long-throw, radial horns, or horns fitted with double drivers. Thus, to achieve high-fidelity results, it is evident that even the best drivers must be driven very conservatively on short, fast flare rate horns, and rolled off at between 7 and 10 kHz into "real" tweeters.

The requirements for midrange amplifiers are high slew rate and power output, to eliminate slewing and clipping respectively. Slewing causes the midrage to sound "grainy," while clipping causes audible third-order harmonic distortion, especially in drivers that respond to high frequencies. Amplifier clipping also increases the power that must be dissipated by the voice coil, a major source of thermal overload. Compression-type diaphragms are usually destroyed by a combination of over-excursion at the lower crossover point, and overheating.

High-Frequency Range

In any audio reproduction system the most important factor determining quality is speed of the electronics. When an active crossover is used, all this speed appears in only one band: the high-frequency range of 5 to 9 kHz, and above.

The HF band carries the leading edge and high slew information of all the program material. Transformerless electronics have just recently opened up areas of the spectrum previously occupied only by noise. As a result, reproduction of high frequencies at high SPL is a relatively new problem for PA system design. Because of the nature of sound and air, the relationship between frequency and SPL is as follows: to reproduce half the reference frequency (one octave lower) at the same level requires four times the excursion of the tranducer.

Since tweeter excursion is very limited, the spectral content of the tweeters must also be very limited to achieve adequate high-frequency energy. Even though the efficiency of tweeters is usually relatively high, their acoustic power output is small, and large numbers of them often are required in a well-designed PA system, particularly in a flying system where the speakers (tweeters) are further from the audience.

In fact, the number of tweeters required in a system can easily be several times the number of midrange transducers. Finally, the placement of tweeters to provide a dispersion similar to the midrange units is a difficult problem. If the tweeters are not arrayed in such a way that they can

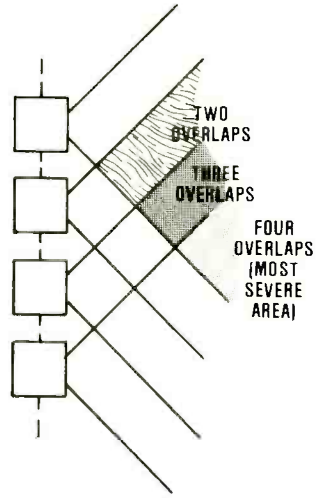


Figure 10: Overlap Patterns with 90-degree Horizontal-pattern Horns.

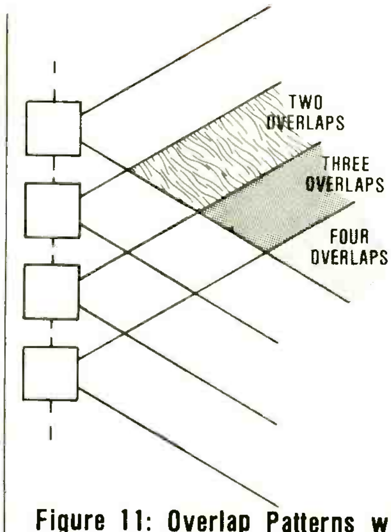
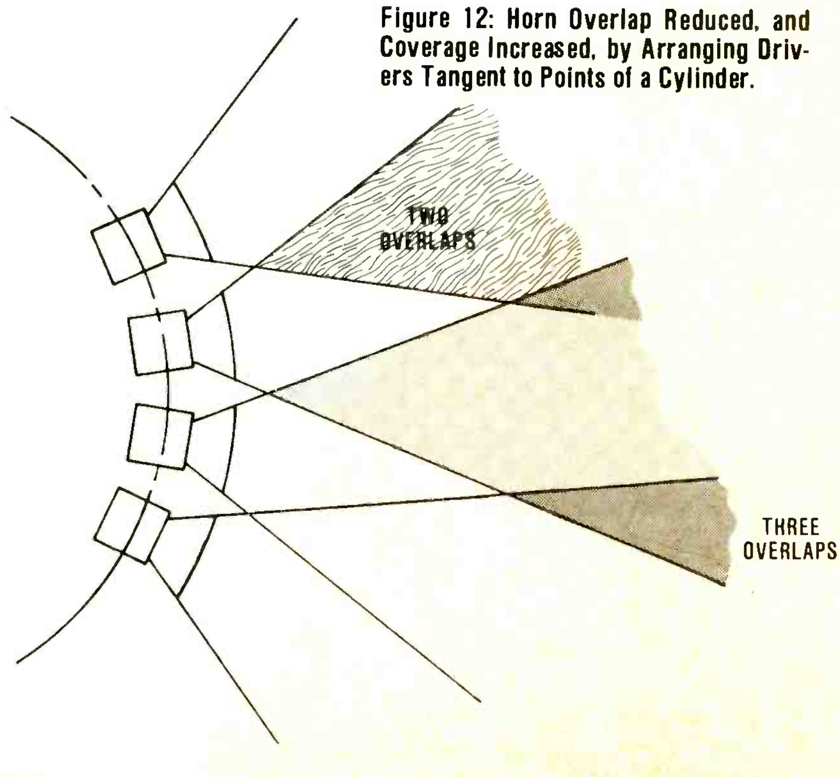


Figure 11: Overlap Patterns with 60-degree Horizontal-pattern Horns.

Figure 12: Horn Overlap Reduced, and Coverage Increased, by Arranging Drivers Tangent to Points of a Cylinder.



couple effectively, the small acoustic power of a single tweeter will not "throw" very far.

Unlike horns, tweeters have no problem with excess horizontal beam width; their problem is that a single linear array does not have sufficient vertical beamwidth for adequate coverage as an indoor flying system. Often several of these arrays must be used.

The amplification required to drive tweeters has very stringent performance parameters: it must have full power and stability at 20 kHz. The long cabling associated with flying systems will make stability more difficult. High damping factor at high frequencies is a fairly good indicator of amplifier stability: DF = 40 at 8 ohms and DF = 80 at 4 ohms are values that produce no detrimental effect on a system's operation. These values may seem low, but at high frequencies the damping factor of some amplifiers that are rated 250 at 100 Hz may not even have a DF of 25 at 10 or 20 kHz.

The amplifier must also be capable of applying much more power to the tweeter than its recommended rating; this is to allow for program dynamics. The HF energy spectrum consists predominately of transients, with very little average or RMS power. Clipping of the amplifiers must be avoided, even though the distortion products generated are beyond audibility, because the power that must be dissipated in the tweeter with a clipped signal is much greater.

The slew rate required of the amplifier is dependent on the slewing speed of the driving electronics. Allowing for 20 dB of voltage gain in the amplifier (being very conservative), it must have 10 times the slew rate of the electronics. With some of the new mixing consoles available for use on the road, the required amplifier slew rate is greater than 50 volts per microseconds. With all of the above considerations in mind, the concept of tweeters driven by the midrange amplifier with a passive crossover seems unsatisfactory.

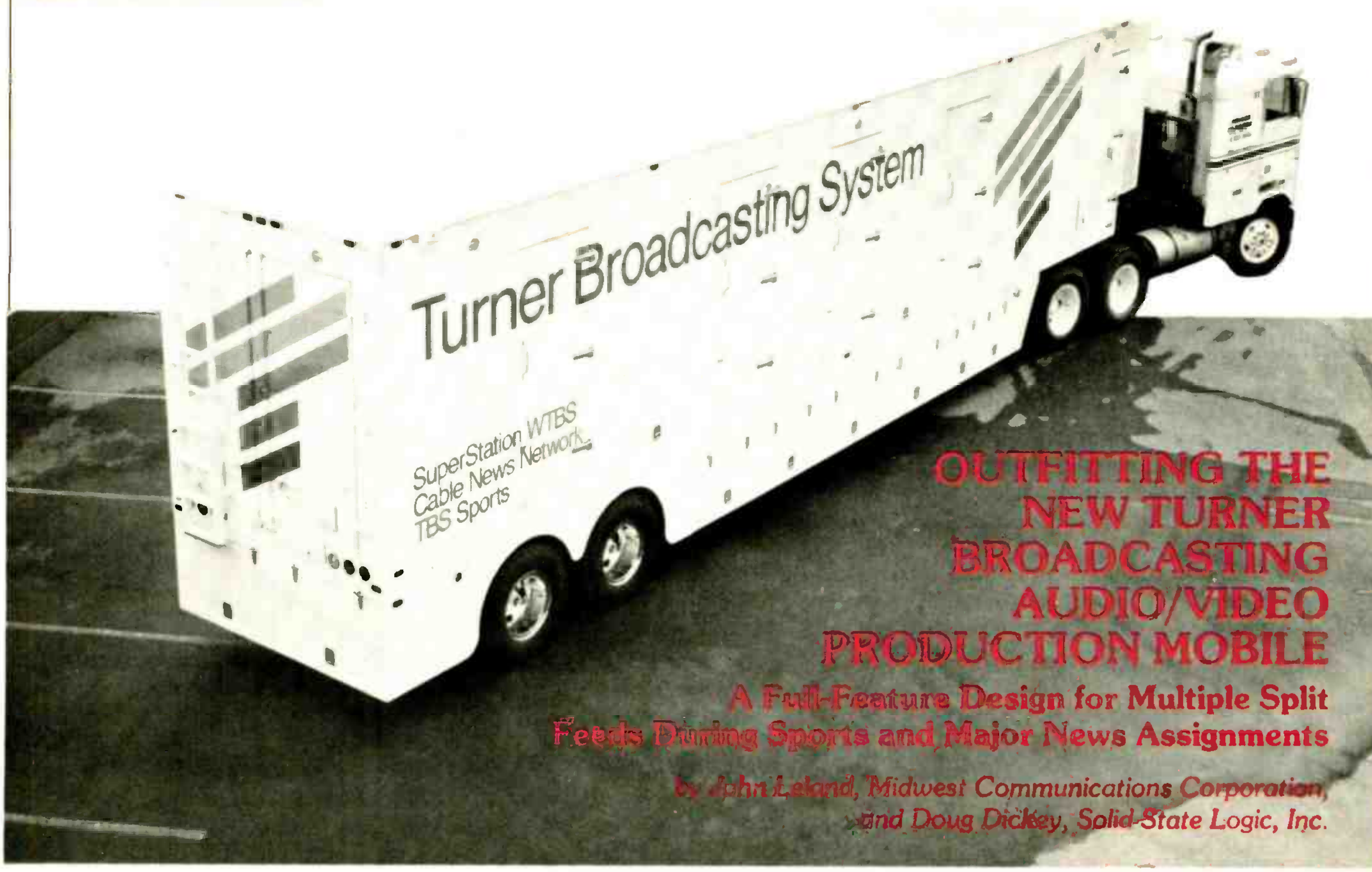
System Headroom

The primary objective in front-end functional design — crossover, limiters, and equalizers — is to process the program source and distribute the output to the power section in such a way that the result is a fully integrated, multiway system. With the advent of active crossovers for use in multiway systems, the power handling capability, frequency response, and distortion of PA systems have been dramatically improved. However, the overall "sound" of the system has often been degraded, resulting in a louder conglomeration of woofers, horns and tweeters that does not perform well as an integrated system.

In addition to dividing the energy spectrum into N sections, the crossover affects the following parameters:

- Overall system damping (transient behavior);
- System headroom;
- Phase response; and
- Amplitude response.

The crossover is the most important piece of electronics in a PA system. Phase alignment and response are generally determined by the order (second, third, etc.), and the type (Bessel, Chebyshev, Butterworth, etc.) of the filters used. While positioning of PA components also contributes to the phase-alignment error, crossover operations dominate this error source. System damping and transient behavior are fixed by the type of filter. Amplitude response and system headroom are both functions of the crossover points selected, and the level of each band sent to the amplifiers. (See accompanying sidebar figures.)



Photography courtesy of Midwest Communications

OUTFITTING THE NEW TURNER BROADCASTING AUDIO/VIDEO PRODUCTION MOBILE

A Full-Feature Design for Multiple Split Feeds During Sports and Major News Assignments

by John Leland, Midwest Communications Corporation, and Doug Dickey, Solid-State Logic, Inc.

Audio used to be relegated to an "Oh, by the way" status as far as television broadcasting was concerned. Several reasons were customarily given why this lack of concern was so: "Everyone listens on four-inch speakers anyway" (and) "There isn't any way to transmit stereo, so why bother with the mix" (or) "Videotape won't record audio worth a darn, so there isn't any point being concerned about it." And so on.

A number of broadcasters, however, have long recognized the need for treating audio as something other than a step child in television broadcasting, and also carry that philosophy over into their remote operations. When, in the spring of 1983, Turner Broadcasting System decided to build a new 45-foot mobile unit, a basic requirement was that it should incorporate full audio facilities. Not just the normal 24 or 32 input/output channels, but instead the maximum technology available to complement the full-service video technology planned for the truck.

SuperStation WTBS Field Operations chief engineer, Joe Wheeler, and Midwest Communications Corporation design engineers began system design in the summer of 1983; at the same time, WTBS Field Operations lead audio engineer, Bob McGee, began a search for the perfect audio console. His search finally took him to the AES Convention in the fall of 1983, and his final selection was made at the show: a Solid State Logic SL-6000E Series Stereo Video System.

Audio Requirements for Television Remotes

Since the new TBS mobile is intended primarily for live broadcast, multitrack machines were not considered as standard equipment, although they can be fitted if necessary. In addition to sportscasting, the truck is aimed at special-events coverage, such as the recent Democratic and Republican national conventions. This type of programming is largely unformatted, requiring the mixing engineer to have virtually immediate control over a great many sources, which may be called upon in almost random order.

Furthermore, many live events such as these require multiple audio feeds, consisting of a main program mix, and a variety of mix-minuses or splits. The Turner Broadcasting System, through its various divisions of WTBS, Cable News Network, and TBS Sports, has a strong commitment to stereo television, and it was deemed necessary to provide stereo for the splits as well as for the main program. Recognizing that anything but the highest quality audio would defeat the benefits of stereo sound, another requirement was for short signal paths.

Because special-events coverage frequently involves less than ideal control at the source, Turner's engineers also wanted as much corrective signal processing power as could be accommodated within the space constraints of the vehicle. Indeed, space was a major concern as were reliability and ruggedness.

After careful evaluation, Turner decided on a Model SL-6032E Series Stereo Video System, which is just under 72 inches wide and 48 inches deep, placing all controls within reach of a single, centrally positioned engineer. The remote patchbay required 9U of 19-inch racking; an additional 16U holds the main and backup power supplies with their instantaneous changeover unit, and an SSL Dual Microphone Pre-amplifier rack, which houses up to 24 remote mike pre-amps with full control and level indication.

The console is equipped with 32 input/output modules, each controlling two signal paths. The main path is fitted with a 104mm Penny & Giles fader controlling the channel VCA. The secondary path is fitted with a 65mm P&G audio fader; in multitrack use, this serves as the track monitor. In the typical Turner application, however, these smaller faders are interfaced with the remote SSL mike pre-amps. The result is something like a dual manual organ, with two rows of faders, one above the other, with

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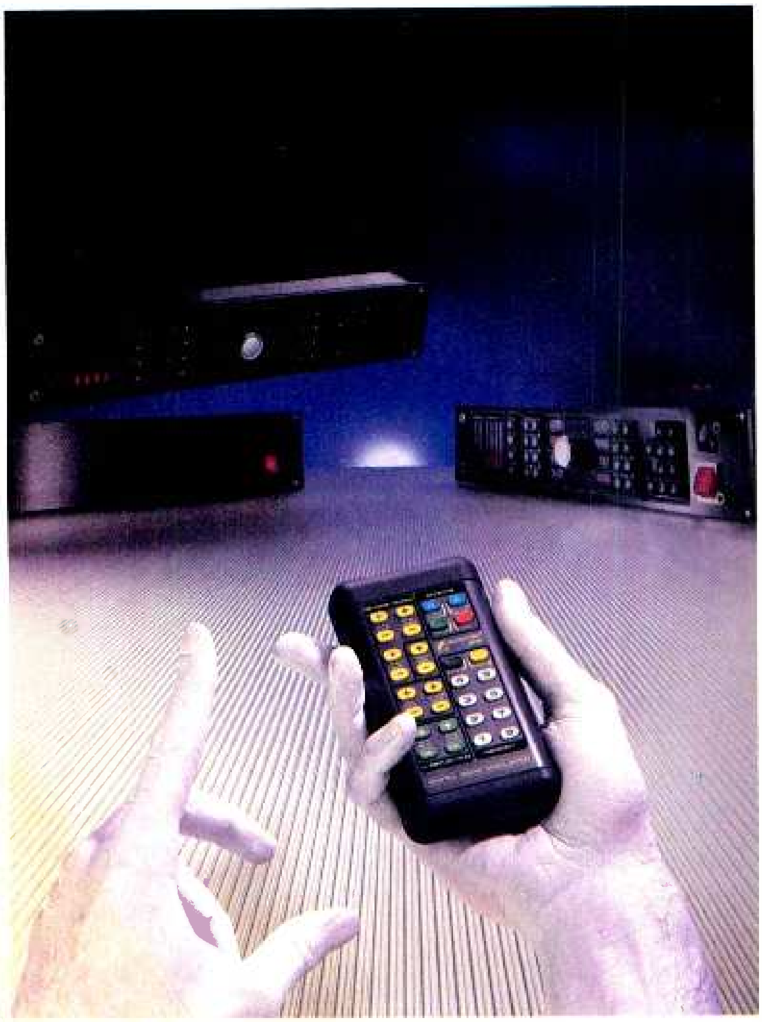

Programs

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Density of resonance	Max 3 per Hz of bondwidth depending on room size
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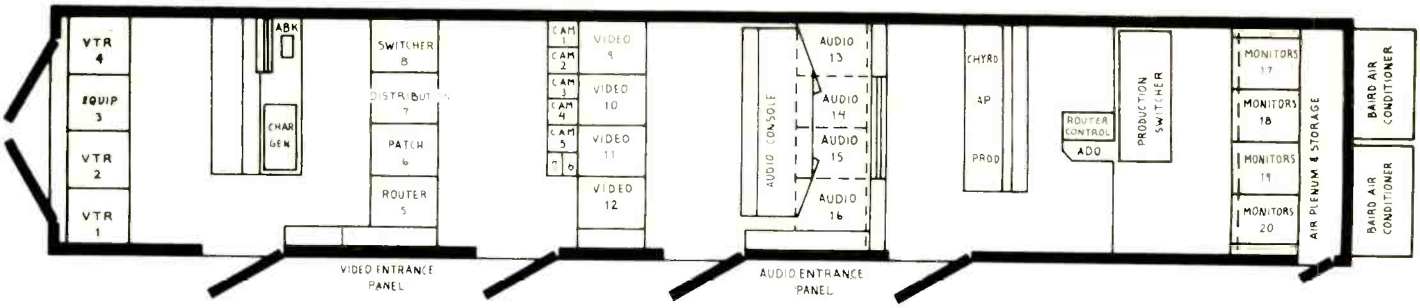
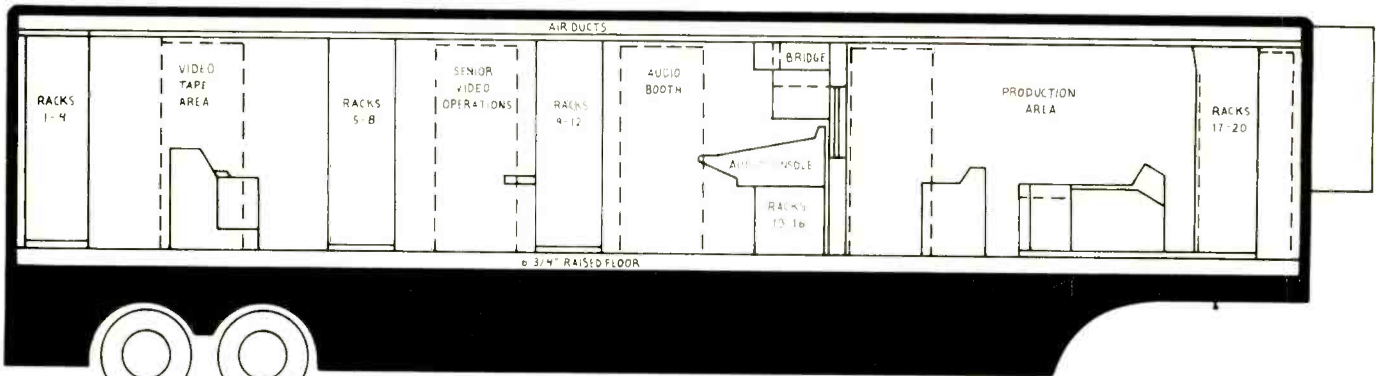


FLOORPLAN AND LAYOUT OF TBS AUDIO/VIDEO MOBILE VEHICLE



TURNER BROADCASTING

fader, panning and assign control over a possible total of 64 simultaneous mike or line inputs. In addition there are four stereo effects returns.

Built-in to each of the 32 I/O modules is a four-band parametric equalizer, continuously variable (18 dB per octave) highpass and (12 dB per octave) low pass filters, plus a full-feature compressor/limiter and expander/gate. These units are normally out of circuit, but may be switched into either the large or small fader signal paths in different orders. Such pushbutton signal-processor routing allows the engineer to structure over two dozen useful variations of the audio paths through each module.

Overall signal flow is determined by the Console Master Logic which offers six "standard" operating modes. One of the console's most powerful features for live broadcast is the ability to further amend these paths using the patch-free subgrouping facilities in each module.

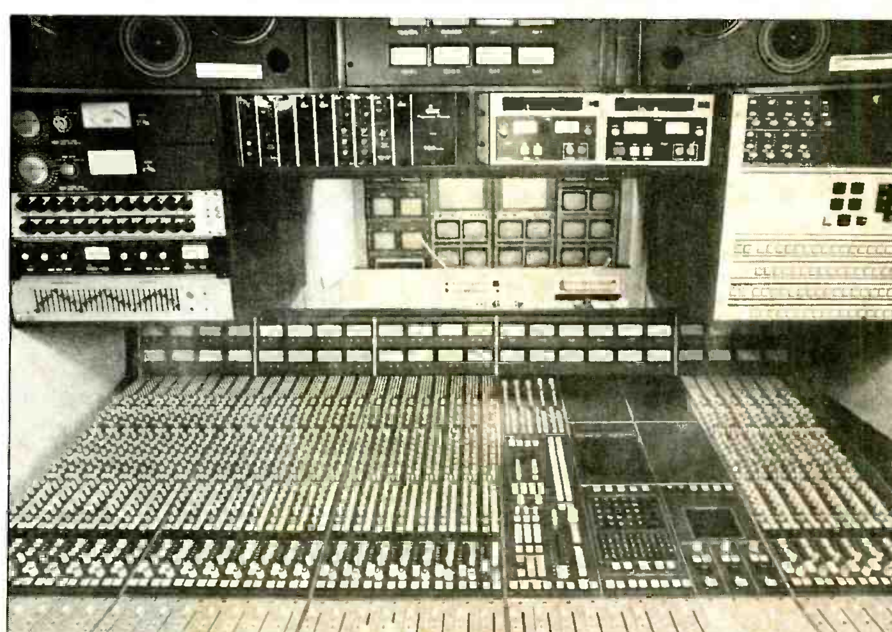
Subgrouping Flexibility

Essentially, any of the 64 faders in Turner's SSL can be selected as an audio submaster for any other faders. As many as 32 such groups can be created using a simple pushbutton scheme. In addition, any of the 32 large faders may be assigned to one of eight VCA Group Masters, which are controlled by dedicated faders at the console's center section. This scheme allows each operator to create his or her own logical flow configuration, tailored exactly to suit each specific production.

For example, all of the audience and playing-field microphones being used at a sportscast can be set up on small faders at one end of the console. Once their "interior" balance has been set, the audience mix can be routed to a large fader that will serve as the audio subgroup master. The playing-field mix can be routed to another audio subgroup. If there is a live band, their balance might also be set up on small faders and routed to a large fader serving as yet another audio subgroup master. (This process may be accomplished in stereo if desired.)

Announcers, commentators and field reporters microphones, which may be frequently juggled during the program, would be brought up on the most conveniently located large faders. Pre-recorded sources would prob-

View from Audio Control Area to forward Video Production Area, with 32-input Solid State Logic SL-6032E console.

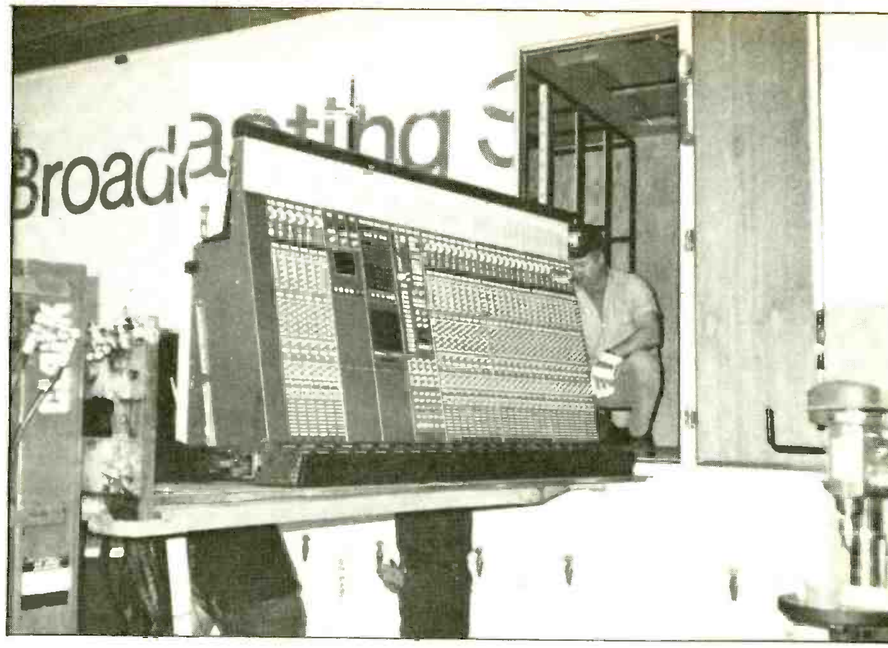


TURNER BROADCASTING

ably also be brought up on large faders.

Signal processors can be switched in where needed. The audio subgroup masters provide a quick way to derive foldback from multimike sources, as well as overall compression, EQ, and the like for each major constituent of the program mix. To make the broadcast even more manageable, the individual and audio subgroup faders can then be selected to VCA Control Groups that provide level and cut control only, and have no effect on the audio routing of the signals.

In this example, audience and playing-field microphones could be assigned to VCA Control Group #1, which might be labelled "Ambience." The live band might be assigned to VCA Group #2, labelled "Band." If desired, we could then assign VCA Group Faders 1 and 2 to VCA Group #7, which we could call "Stadium." Similarly, all announce mikes could be assigned to VCA Group #3, commentary mikes to #4, and field reporters to #5. We could then assign VCA Groups Faders 3, 4 and 5 to VCA Group #6, which we could label "Talent." Finally, all of the pre-recorded sources could be assigned to VCA Control Group #8, which we would label "Tape."



The six- by four-foot SSL console during installation into the new WTBS mobile production vehicle. Midwest Corporation reports using the services of a "heavily insured" moving company for the job.

During an actual broadcast, this configuration gives the mixer a cluster of single faders to lift for Ambience, Band, Announce Booth, Commentary Studio, and Field Reporters. It also provides a single fader to control the overall level of all "outside" sound (Stadium); a single fader to control all "inside" sound (Talent); and a single fader to control all pre-recorded sound (Tape).

The beauty of this scheme is that the mechanics of a live mix are reduced to the simplest possible elements, but the security of instant access to each individual source is maintained at all times. Furthermore, as the sources progress downstream, each successively important level of grouping brings their control closer to the central mixing position, eventually resting directly in front of the operator. Finally, the various types and levels of grouping are completely free-form; the operator may essentially re-design the console, entirely by pushbuttons, to suit their own approach to each different project.

Triple Stereo Mix Busses for Split Feeds

Another feature of the SL-6000E Series which is said to appeal enormously to the TBS Remote Department is the provision of three stereo mix busses, designated A, B and C. Pan controls for each of the large and small faders can be assigned to pan between the left and right channels of any of the three stereo Mix Busses. Any or all of the mix busses may then be assigned to the main console output, which is designated "Program." The three stereo busses may also be assigned to each other in various combinations. This may sound a bit confusing at first, but it is actually quite simple — and it is key to the

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

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

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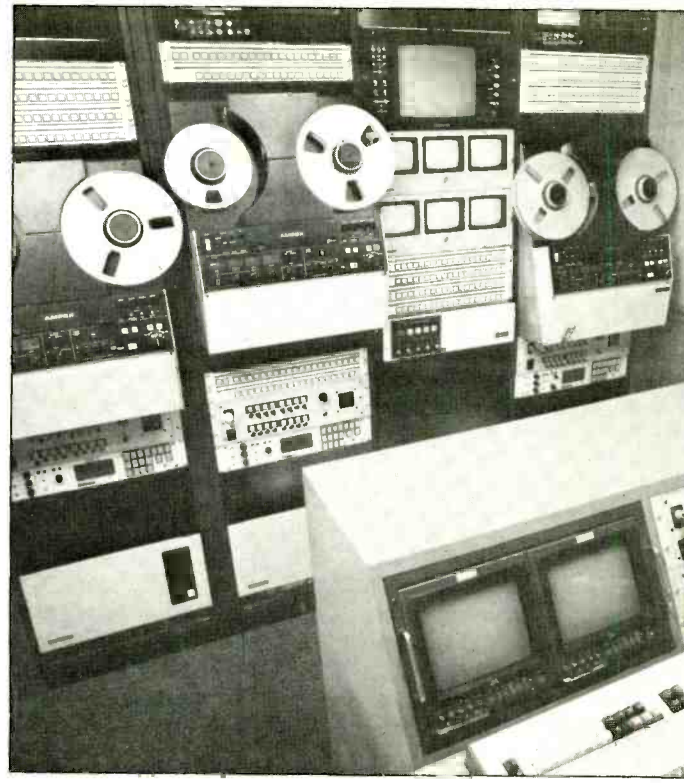


The Video Production Area (left) house a Grass Valley 300-A-3 Series video switcher, and an Ampex ADO Digital Video Effects Unit; the Videotape/Chyron Graphics Generation Area (right) serves as home for Ampex VPR-2B one-inch, C-Type videotape machines, with full timecode and Slo-Mo capabilities.

rapid creation of the stereo split feeds that the Turner organization required.

Controls for the three stereo mix busses are located on a central panel — the SL688V Mix Matrix — located to the right of the eight VCA Control Groups and Master Program Fader. Each of the A, B and C busses has a button that assigns its mix to the Program bus; these are labelled "A to Program," "B to Program" and "C to Program." For standard stereo operation, all faders are assigned to the Stereo A Mix Bus, and the "A to Program" button selected.

The creation of mix-minus feeds or splits is equally straightforward. From our earlier example, we might assign all of the Talent mikes to Stereo Mix A; Ambience and Band subgroups to Stereo Mix B; and pre-recorded sources to Stereo Mix C. Each of the three Stereo Mix busses can be assigned to either or both of the other Stereo Mix busses, using six buttons labelled "A to B," "A to C," "B to A," and so on. These buttons are logically positioned above each of the three "(Mix) to Program" buttons, and enable a variety of feeds to be created. As a simple example, we could assign all of the "Talent" channels to Stereo Mix A; all of the "Stadium" subgroups to Stereo Mix B; and all of the pre-recorded "Tape" channels to Stereo Mix C.

By assigning A to Program, B to C, and C to A, we have created a Program Mix consisting of Talent, Stadium and Tape. We have simultaneously created a feed consisting of only the Stadium sources, which can be used as the "pool" feed to other broadcasters that are going to add foreign language commentary; *and* a feed consisting of Stadium and Tape sources, as a pool feed to same-language broadcasters that may use our pre-recorded interviews, but otherwise want their own announcers.

This simple example only suggests the overall output distribution flexibility of the SL688V Mix Matrix; it is also possible to add a stereo line input to each of the three Stereo Mix Busses. Depending on how we choose to set up our audio subgroups, these additional inputs can be used to create still further subdivisions from which our Program Mix and splits may be derived. It is worth pointing out that we can combine the Stereo Mixes into the Program Bus at unity gain, or insert a stereo fader to provide a final downstream level control prior to the Master Fader.

Each of the Stereo Mix Busses has its own Mono (L+R) feed that follows a separate mono trim control, enabling adjustment of the mono composite to account for differences in relative levels that occur when wide left/right spread of the ambience and music sources are combined with the mostly center information from the talent mikes. A mono Program feed is also always available. Finally, a stereo compressor that acts directly on the Master Program Fader VCAs may be switched into circuit, giving the location mixer an opportunity to control overall dynamic range of the principle mix under his control, rather than encouraging someone further down the line to mangle it.

The SSL console system is supported by dual redundant power supplies, with silent instant changeover in the event of a rail failure in either supply. The master logic is actually fully distributed to prevent a central catastrophe. Each group of eight faders is on its own breaker, and modules are protected by individual fusing and current-limiting resistors. The I/O module motherboards are entirely passive, with all active elements placed on plug in daughter cards with gold edge conector, each secured by a screw. Switching any of the signal processors out of the circuit creates a hardwired by-pass around any trouble, and the frame is designed to allow modules to be "hot-plugged" if necessary. An extensive patchbay provides further crisis-avoidance capability.

Related Audio Facilities

Audio-related capabilities of the new TBS mobile unit do not stop with the SSL board, however, an extensive intercom and IFB system allows production staff to talk with the talent, each other, Telco, PL drops, two-way radio, and engineering from one central control unit. There are also intercom stations provided at entrance panels on the side of the truck, for coordination during setup and tear down activities, plus a separate engineering channel for troubleshooting problems off-line while the shoot is in progress.

The entrance panels are considered unique in that special quality control facilities have been placed in both the video and audio panels. In the case of the video panel, a color monitor and oscilloscope are permanently installed, as well as a routine switcher. This set up allows the engineer in charge to check any video source without entering the truck; since the truck is always crowded during a telecast, such a capability is a real plus. The audio entrance panel contains a routing switcher and line amp for the same purpose. Additionally, the audio panel contains XLR and XLJ connectors wired in parallel, as well as binding posts connected in parallel with the XLs, which effectively eliminates the need for audio "turnarounds."

Multiple wireless microphone and a full complement of shotgun and special purpose mikes travel with the truck.

A Utah Scientific audio routing switcher with a 50×20 matrix allows audio sources to be called up at any production point within the truck, and an X-Y controller allows the semi-permanent assignment of audio functions where required. In addition, should it ever be necessary, bus outputs from the SSL console can be interfaced with off-board multitrack machines, through multipin connectors located at the audio entrance panel.

Video Capabilities

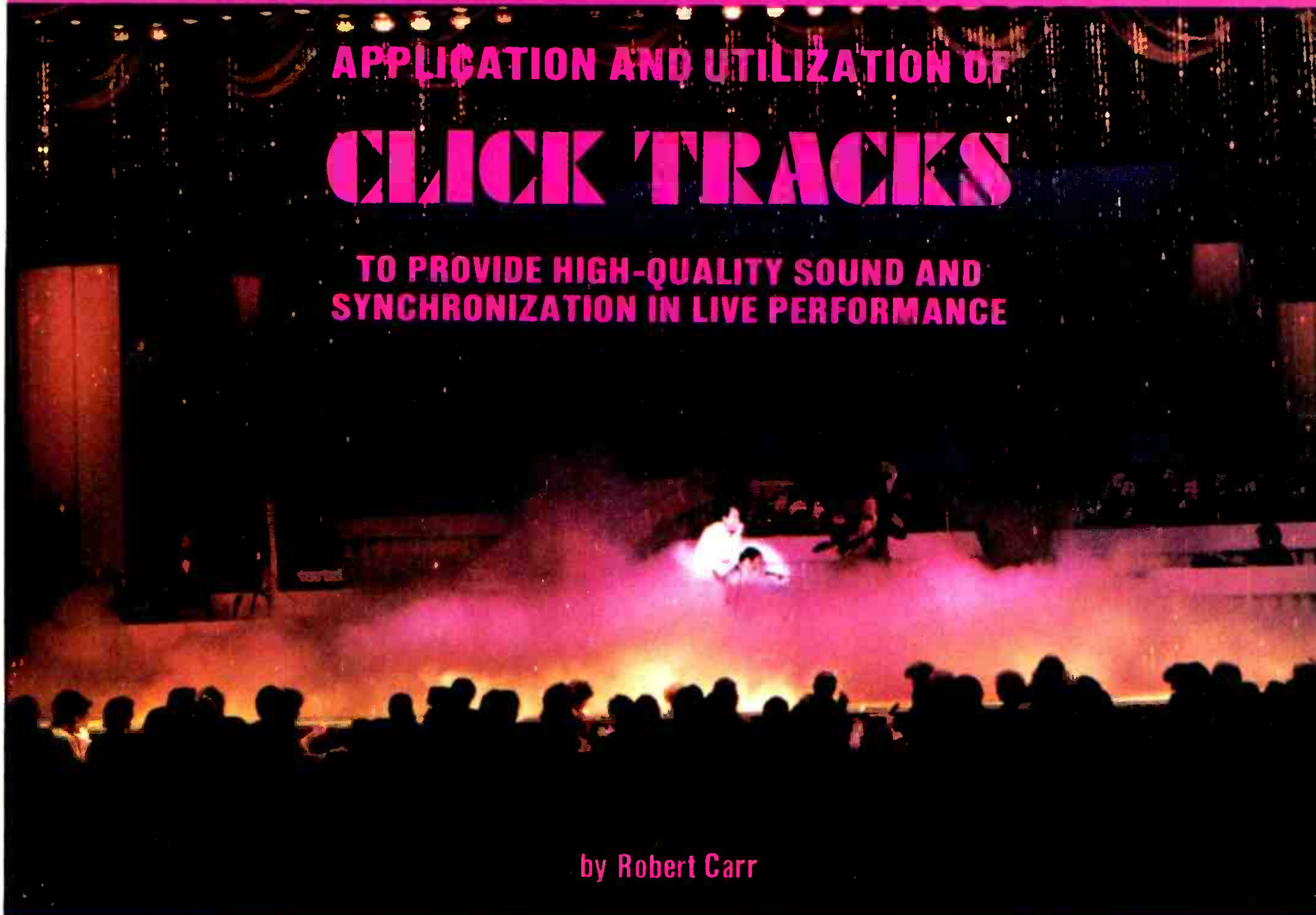
The unit normally travels with five Ikegami HK-357AT Triax computer cameras equipped with 44:1 zoom lenses. Also included in the camera complement are two Ikegami HL-79 hand-held cameras, with Triax adaptors and 17:1 lenses.

A Grass Valley Group Model 300 A-3 Series video switcher has three Mix/Effects banks, and all available patterns and wipe capabilities. Combined with the Model 300 switcher is an Ampex ADO Digital Video Effects unit. (The ADO is generally regarded as *the* premier DVE unit, and it is rare to find one in a remote truck.)

Videotape facilities comprise three Ampex VPR-2B one-inch, C-Type transports, all of which have slow-motion playback options and timecode record/play capabilities. Picture storage is also accomplished on an ABEKAS still-store.

As the accompanying floor plan shows, operating areas are grouped into four rooms: the video production area; an audio area that can view the video area through a Lexan window as required; a video control/transmission area where all engineering functions are performed; and finally the VTR/Graphics area that contains the Ampex video machines, ABEKAS still-store and a Chyron 4100 Series graphics generator.

Construction of the TBS remote truck began in late 1983 at the Midwest Communications Corporation facility in Edgewood, Kentucky, and was delivered to WTBS at the end of May 1984. After testing, the unit reportedly hit the street running and performed 14 baseball games plus two political conventions in San Francisco and Dallas with no major problems. The design was a community effort between WTBS Field Operations and Midwest, and the result is a mobile unit that to many of those involved redefines the phrase "State-of-the-Art." In their advertising Midwest calls the new unit "Supertruck," and Turner Broadcasting System refers to it as "World Class." Neither company is off the mark. ■■■



APPLICATION AND UTILIZATION OF CLICK TRACKS

TO PROVIDE HIGH-QUALITY SOUND AND SYNCHRONIZATION IN LIVE PERFORMANCE

by Robert Carr

Photography by Keith Hubbell

Most forms of entertainment have reached a pinnacle of sophistication that, to a certain degree, has jaded the expectations of the viewing and listening public. The use of dazzling recording techniques and highly advanced equipment can result in music albums that make the artist sound larger than life. The television medium, while normally not accompanied by the most evolved sound quality, has accustomed viewers to specacular, full-color productions with sophisticated, prerecorded soundtracks.

The boundary between what can and can't be done in a live performance has been radically obscured in the minds of the "average" fan, and graphic representations of "fantasy" are now casually taken for granted. The audience doesn't care how difficult a production process is; all they want and expect is the *effect*. The prevailing attitude is one of: "Artists can do it on record, on TV, on film; why can't they do it live?"

Faced with the task of satisfying this level of audience awareness, performing acts and their audio engineers have turned to prerecorded tapes as a means of satisfying the demanding crowd that are asked to spend substantial amounts of money for a night with their favorite entertainer.

To date, only a few rock groups have attempted to augment their shows with prerecorded tracks in an effort to duplicate complex studio techniques on stage. Generally, the use of tape machines as a regular, yet discrete, addition to the stage performance seems to be more popular among television and film people, including such notables as Shirley MacLaine, Paul Anka, Ann-Margret, Cher, Engelbert Humperdinck, and other versatile stars of their calibre. Such acts are primarily production-number oriented, where dancing and theatrics are large components of the overall performance.

In essence, a click-track system is normally considered to comprise of at least two tape tracks, one of which will be prerecorded instruments, vocal and/ or solos with an accompanying musical tempo and timing clicks. Their use affords the opportunity to enhance a show with otherwise cost-prohibitive orchestrations, and/or with a performance consistency probably unattainable any other way.

A-1 Audio, based in Hollywood, is described as supplying the majority of click-track systems for live sound-reinforcement applications in Las Vegas-style showrooms, and during concert tours. Utilizing its extensive in-house facilities, A-1 designs and builds fail-safe, click-track systems that can withstand the punishment of non-stop touring.

Al Siniscal, owner of A-1 Audio, and himself a veteran sound engineer, is said to have perfected the click-track idea, while working with the Paul Anka show in the early Seventies. "Live music shows are becoming more theatrical," he says. "For example, Engelbert Humperdinck now does a sketch that includes a dialog with his conscience. The sound effects and


And now a message on Yamaha's new RM1608 recording mixer.
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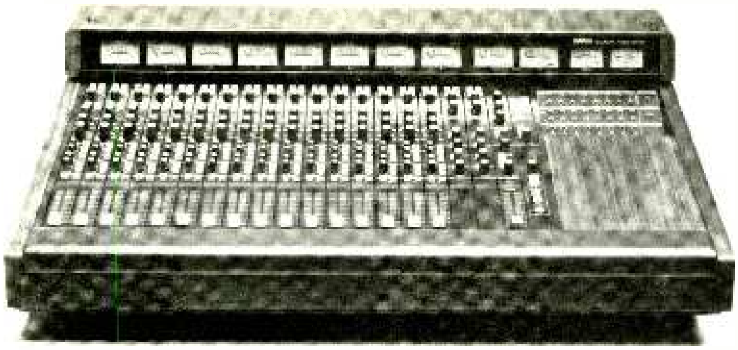


RM1608

SPECIFICATIONS

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

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

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

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

MAXIMUM VOLTAGE GAIN (INPUT GAIN "–60")

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

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

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

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

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

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

Hum and Noise are measured with a –6dB/octave filter at 12.47kHz; equivalent to a 20 kHz filter with infinite dB/octave attenuation.
*0dB is referenced to 0.775V RMS.
• Sensitivity is the lowest level that will produce an output of –10dB (245mV), or the nominal output level when the unit is set to maximum gain.
• All specifications subject to change without notice.

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

And in the tradition of Yamaha's sound reinforcement mixers, the RM1608 sets new standards of reliability as well as ease of operation. For complete information, write: Yamaha International Corporation, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Ave., Scarborough, Ont. M1S 3R1.

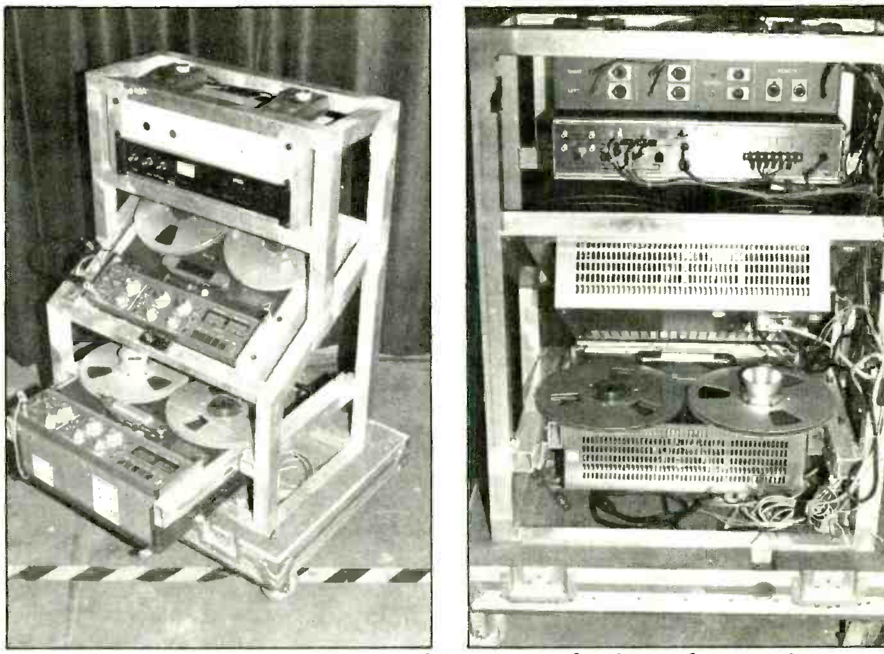
YAMAHA

For additional information circle #100

the responses to his questions, which he asks live during the show, all have to be prerecorded and played back at the appropriate times. The tape machine is started, stopped and recued maybe 25 times so each answer starts correctly."

Similarly, according to Siniscal, Johnny Mathis' hit recording with Dionne Warwick could be done live with a substitute vocalist supplying the female lead. Instead Mathis chose to have Warwick's original vocal transferred to a click-track tape system to accurately duplicate the original single on stage.

In both these instances, of course, the audience knows that the second voice is replayed from tape. But in most situations the process of incorporting tape-recorded segments must remain invisible to the listeners, in order for the illusion of a totally live performance to be maintained.



Front and rear views of custom-designed carrying frame to hold two Revox B77 two-tracks, and companion dbx Model 157 noise-reduction unit.

SAMPLE APPLICATIONS FOR CLICK TRACKS

To quote an old cliche: "The only boundaries are the limits of your imagination." And so it is with prerecorded tape and click-track systems. For any number of reasons, practically anything that can be recorded may be added to a show via tape. Program material can range from simple hand claps and knee slaps, to complete vocal and/or instrumental orchestrations. However, the amount of use during a given show usually depends on such variables as available space on stage for musicians, budget, complexity of songs or effects, stamina of the performer, and so on. Some of the more common applications are:

- *To provide a bridge or a rest for the performer so the vocals can continue while the singers are doing something — such as strenuous dancing — that makes singing unfeasible.*

"When a performer is being carried on the backs of eight dancers, then flipped into the air, and spun three times, singing or even miking the singers during that kind of movement is difficult, if not impossible," says Siniscal. "Because some continuation of the music needs to occur, be it either the lead vocal or just 'oohs' and 'aahs' for the background parts, the engineer can switch to the click track to supply a good audible version."

- *To provide a solid beat for a complicated dance number.*

The energy level of a show often varies from one performance to the next, depending on the excitement on stage and rapport with the audience. But if a show format calls for elaborate dancing, the music must be played at the same tempo every time. A slower or faster pace in a complicated dance tune could cause the dancers to misstep, and maybe twist an ankle or even break a leg. Prere-

CLICK-TRACK PRODUCTION

Producing a click-track tape for live performance can be as complex as any other multitrack recording session. To prepare tapes for Shirley MacLaine's current show, engineer Ken Newman has his own techniques, outlined as follows:

"Although the music section is quite short relative to the whole show, preparation of the segment took a bit of planning and time to complete. Basically, the group did a small, multitrack recording session. The click was laid down first on track #1 [of the eight-track tape]. Because the song involved some tempo changes, we used a drum machine to generate the clicks [a wood-block sound]. Usually we'd get the click from a really precise metronome, but the drum machine allowed us to vary the tempo easier.

"The entire rhythm section — guitar, piano, bass, and drums — was recorded on track #2, while the band listened to the click tempo through headphones. The singers and the sound effects were laid down on the remaining six tracks, using the composite music bed as a guide. The vocalists never heard the click during the session; they sang along with the music just as they would on a normal overdub session. Likewise, while performing live on stage, the band listens to the click through headphones, and the dancers hear only the music.

"Transfer of the studio multitrack to the two-track real-to-reel was similar to any other two-track mixdown. The click from the multitrack went to one track of the two-track. We did several vocal tracks to fatten up the sound. The vocals and sound effects were collapsed to mono and laid on the second track of the two-track machine. Once the vocal parts were recorded in the studio, the music track was erased, because that was for vocal reference only.

"When you're dealing with anything more than one voice, it's nice to have a separate track for each element: one for lead vocal; one for background vocals; and so on for the other elements, like strings or percussion. An Otari four-or eight-track gives you more control over the individual equalization settings [during remix], so you can really beef up the vocals and blend them together according to what is happening live, or how the sound system is reacting to the room on a particular night.

"By doing the recorded tracks ourselves, we knew what kind of sound to go for so the taped vocals would match the live sound on stage. When we remixed the tapes, we recorded the program pretty hot. The click on the other hand is fairly low [-10 VU] so we don't get crosstalk into the program channels.

"Some people use noise gates to reduce the [crosstalk] noise even further. I prefer to run both tracks through a dbx Model 157 noise-reduction unit. That way there's no hiss all of a sudden when you flip on the tape and bring up the house faders in the middle of a tune."

corded music tracks with clicks provide a solid tempo from show to show.

• *To ensure the consistency of a difficult or demanding vocal or instrumental performance.*

Occasionally, a lead vocalist may find that due to the pressures of performing regularly, they can't sing a particularly high note exactly right every show, or the artist may be out of breath after a dance number.

Also, bands cannot always guarantee that they will play the particularly complicated time and key changes of a new song perfectly every time. The performer and/or band can prerecord the tricky parts, and then mime to the tape. This technique is especially common during complex television productions.

"Paul Anka's song, 'Happier,' started out very slow, and gradually built through several tempo and key changes," Siniscal explains. "If the song was played too fast or too slow, the impact of the tune was ruined."

For the benefit of Anka and the musicians, the bass guitar part and a strong rhythm beat (congas and marachas) were prerecorded on a two-track tape and played through both the house and monitor systems. The instrumental rhythm tracks themselves served to keep the time, so clicks were not needed. The rest of the band played live along with the prerecorded tracks.

• *To duplicate complex studio effects.*

Given sufficient time, practically any instrumental, vocal or miscellaneous sound effect can be produced in a recording studio using multi-tracking techniques and multiple-effects processing. Obviously, one pass in a live performance isn't always sufficient to reproduce a "hit" sound on stage. Pre-recorded tapes allows the house mixing engineer to "dub in" the sound at the right time during the course of a show.

• *To add lush orchestrations to a small rhythm section.*

The outrageous costs of touring usually make the possibility of hiring a large orchestra out of the question. However, if the beautiful string accompaniment, for example, is an integral component of a particularly successful arrangement, the audience expects to hear those same music parts as though the live performance were the original record. Again, a prerecorded tape can be blended with the live music track to create the audible illusion that a 40-piece string ensemble is present.

Engineer Ken Newman at the console in New York's Gershwin Theater during a Shirley MacLaine show.

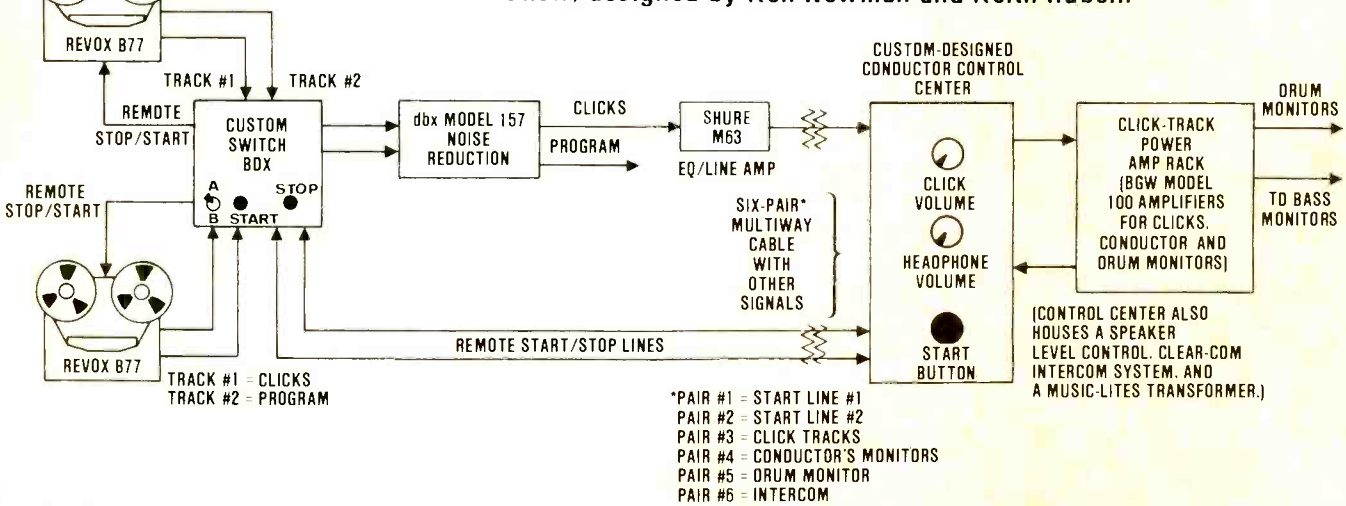
CASE HISTORIES OF SPECIFIC APPLICATIONS

According to her house-mix engineer, Ken Newman, Academy Award winner **Shirley MacLaine** has often turned to click-track recordings to enhance some of her stage show material. Currently, tapes are necessary in only one song—choreographer Bob Fosse's arrangement of "Sweet Georgia Brown."

"We're using the click to provide a good-quality vocal track during the dancing," says Newman. "If the four dancers and Shirley were miked for that segment, the quality would not be as good as the rest of the show. And because they are all wearing rather skimpy costumes, miking them inconspicuously with wireless units would be pretty tough. The taped program track replaces Shirley's and the dancers' voices, so they don't have to actually sing while they're dancing.

. . . continued overleaf —

Signal Flow of Click-Track System for Shirley MacLaine Show, designed by Ken Newman and Keith Hubell.



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For additional information circle #101

In addition, the tape also includes sound effects like finger snaps and knee slaps."

Newman usually positions the tape machines right next to him at the house console. But for the New York engagement at the Gershwin Theater during May and June of this year, the decks have been placed backstage.

"I have very little room in the audience and people are all around me," he says. "With the machines out of sight, nobody knows they're running. They all think the vocal parts are totally live, especially since other parts of the show are miked with body packs."

The illusion of "live"-ness is aided by the low-profile Sennheiser stethescope headphones that the band uses on stage. Prior to playing the prerecorded, two-track tape (clicks on track #1 and program material on track #2), the musicians and conductor inconspicuously slip on the headsets, which are powered by a rack of BGW Model 100 amplifiers. Because the program tracks are played through the house and monitor system, the headphone lines carry only click information. A dbx Model 157 noise-reduction unit and a Shure M-63 Audio Master mixer provides the conductor with some control over the overall level and EQ of the click track. Each musician has an individual volume control to adjust the loudness of his or her headset.

When everybody is ready, the conductor hits the remote "start" button connected to a pair of Revox B77 two-tracks that have been aligned before the show to be as "on-speed" as possible, and cued to the beginning of the count-off at the head of each tape. As soon as the button is pressed, the count-off is heard immediately, followed by the song. At the end of the taped segment, a small strip of clear leader trips the Revox end-of-tape sensors, and both machines stop automatically.

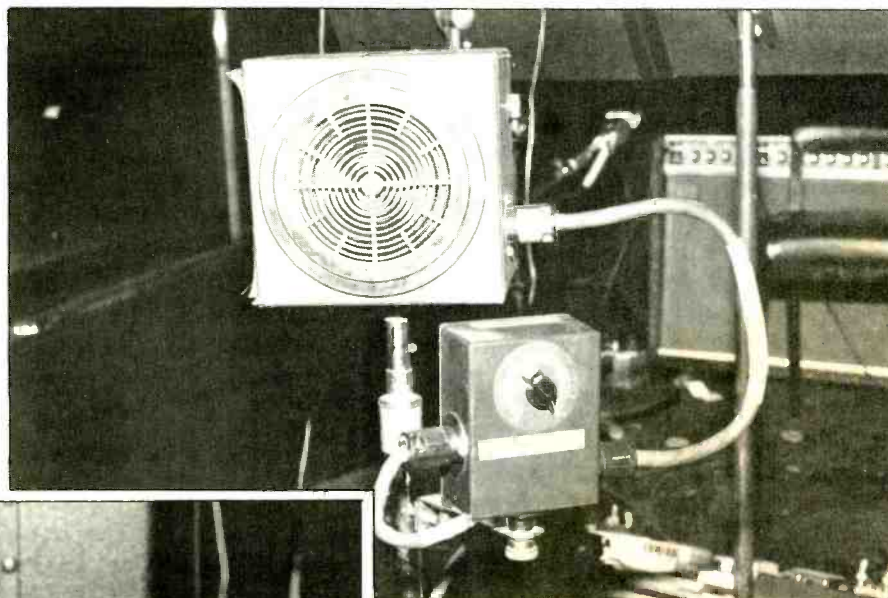
"The system has proven extremely reliable," says Newman. "The automated stop feature serves as an extra hand in case nobody can get to the recorder after the song has finished."

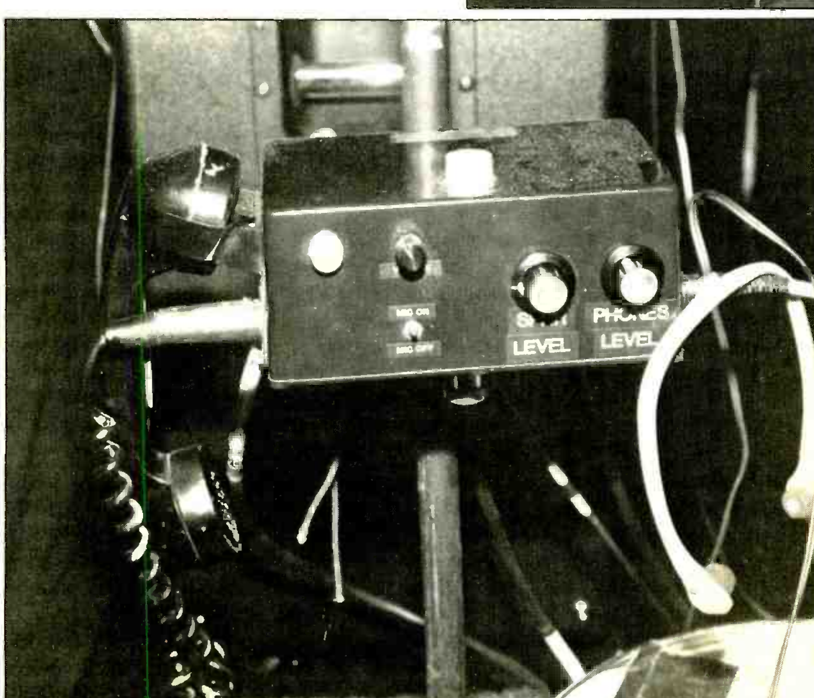
The tape signal is treated like any other input sent down the snake to the house console. Because the two machines are used for only one song per show, and there's no need to recue, their position backstage has reduced the number of lines to just one: the program material that plays through the house system. The remote-start lines for the conductor, the click signal for the headphones, and the program feed for the monitor mixer all remain on stage, thus freeing up several more lines in the snake cable.

As with Shirley MacLaine, the click-system program tracks for the **Ann-Margret** show supply the lead and/or background vocal parts during the dance sections. However, Ann-Margret's click system plays a more comprehensive role in the overall live presentation. Projection equipment and laser effects are also cued via tape. Utilizing two Revox B77 two-tracks, the click and/or stereo or mono program information is sent to the lasers and projectors operators, who use the signals as cues to trigger special visual effects. Occasionally, timecode information and program tracks are interfaced with the effects generators for direct control. (The operators can override these timecode signals if necessary for spontaneous use of lasers and projectors.)

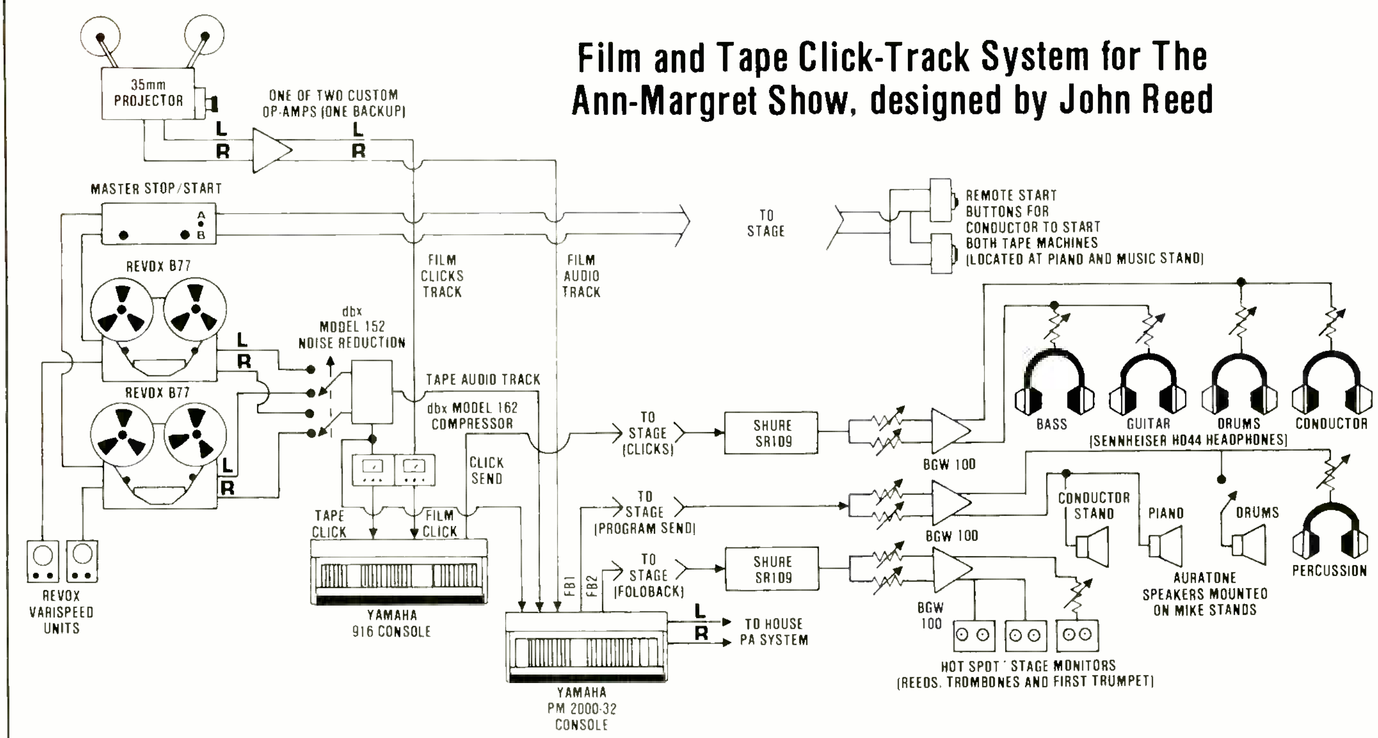
As with instrumental and vocal program material, the band members and conductor hear the click through Sennheiser headphones, so that the effects stay in sync with the song. The conductor has a miniature mixer and amplifier setup at his station, and via these controls he can set amplifier levels to the guitar player, the drummer, the bass player and him-

Headphone control for the drummer includes an intercom to monitor and house engineers, plus conductor; shown below left is an Auratone monitor speaker fitted with heavy-duty audio connectors.





Film and Tape Click-Track System for The Ann-Margret Show, designed by John Reed



CLICK TRACKS

self. The clicks are compressed, too, with a dbx Model 160 at the house console. "The clicks produce strong transients that tend to overload the brain information," explains house engineer John Reed. "The compression gives us a louder level and better intelligibility without the overload."

In addition to individual volume controls featured on each headset, certain members of the band have their own Auratone speakers through which program material is played for reference and cues. The conductor also has an LED display that lights up for every click. In that way, he has visual as well as aural contact with the taped tempo.

A remote start button provides the conductor with more control over the show. Because he is located on stage, the conductor knows when Ann-Margret wants to keep talking to the audience or, conversely, eliminate the rap and get right to the music. Or depending on audience response, maybe eliminate the next song completely.

"The setup is really a safety precaution," explains Read. "The conductor is in direct contact with the artist, who may want to alter the program. I have a communication line with him just in case I have to recue another song at the last minute."

For Ann-Margret's show, where several tunes are prerecorded, the B77's automatic-stop function via

ANATOMY AND OPERATION OF CLICK-TRACK SYSTEMS

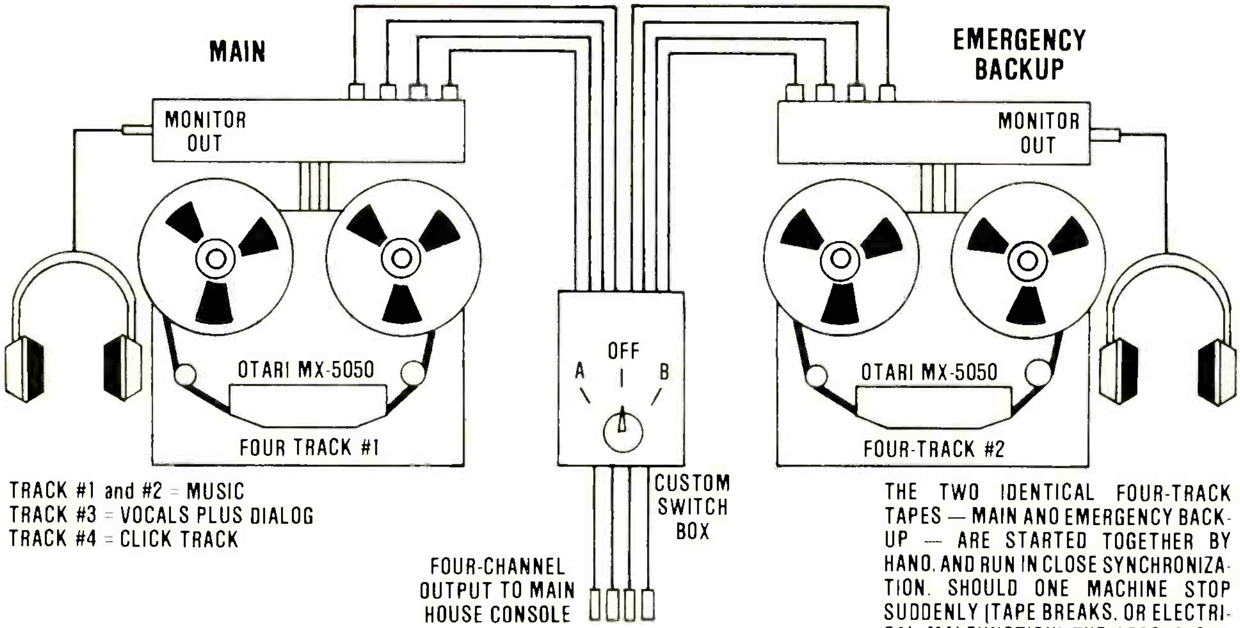
Traditional pre-recorded tape and click-track systems are usually based around two-track reel-to-reel machines, although four and eight-track decks may be used for more complex productions, and to increase the processing flexibility of separate program tracks. One of the two tracks holds the program material that augments the show, and plays back through the house speakers and stage monitors. The program, which may be just the lead singer's vocal track, special effects, or several instruments, such as strings or a horn section, is treated like any other channel in a live mix, being sent through a single channel (or channels for stereo mixes) of each control board.

On the second track of the two-track tape is the click itself, which is generally played back through headphones to the conductor and specific band members, such as the drummer, bass player, and sometimes the piano player. Other people occasionally receive a click feed, although the rhythm section is most important since it sets the tempo for everybody else.

Recorded Click-track Level and Reference Tones

One of the most important aspects of preparing a click track is the level at which it is

Pair of Revox B77 two-tracks with varispeed controllers for each deck



THE TWO IDENTICAL FOUR-TRACK TAPES — MAIN AND EMERGENCY BACK-UP — ARE STARTED TOGETHER BY HAND, AND RUN IN CLOSE SYNCHRONIZATION. SHOULD ONE MACHINE STOP SUDDENLY (TAPE BREAKS, OR ELECTRICAL MALFUNCTION) THE SECOND CAN BE QUICKLY SWITCHED IN AT THE CORRECT POINT IN THE SHOW.

Double Four-Track System for The Cher Show, designed by Andy McCartney

OPERATION OF CLICK-TRACK SYSTEMS — continued . . .

recorded. A normal recording level of +4 dB can cause the clicks — especially those derived from more inexpensive click units — to cross-talk into other program channels, and then leak into the house PA system. A level of -10 VU has proved to be sufficient for cue purposes, while ensuring that the live performance isn't disturbed by annoying leakage. (It goes without saying that clicks are never routed through the monitors, since they can be picked up by the mikes and sent right out into the audience.)

Every click track should contain a reference tone (such as an A440) recorded at the head of the tape. By using a piano or tuning device, the speed of the tape machine can be reset precisely before every show so that the program material is at the correct pitch, and matched with the instruments on stage. If this isn't done, the vocal track may be out of tune when the tape program is switched into circuit. Additional reference tones of 100 Hz, 1 kHz, and 10 kHz are handy when aligning the electronics of the machine prior to use.

Tape Leaders and End-of-tape Sensors

Click-track tapes should be properly leadered. In most cases, about four-inches of leader is sufficient between each prerecorded segment of the tape. If the leader is much longer, quick cues and changes in song sequence during the show become difficult to implement. The name and duration of the following song should be written on the leader to ensure that the operator doesn't cue up the wrong tune. Start marks are also a good idea, so that the engineer knows where to set the tape when he recues it after each song.

It is often recommended that tape machines with automatic "shut-off" devices be used. For example, the Revox B77 transport features an LED and photo-sensor that automatically stops the machine when clear leader passes between them. In this way there is no possibility of the tape running on to the next track, and playing inappropriately — which is especially important if the engineer is busy operating a mixing console, and doesn't have time to reach over and shut off a tape deck at the end of a song. If the tape runs over into the next sequence, the audience discovers that the show is on tape; the performer gets embarrassed, and the magic of the show is ruined.

Back-up Tape Systems

In shows where only one or two songs are prerecorded on tape, or where the performer could sing the song just as well if something happens to the prerecorded version, one machine is probably sufficient. Two tape machines running simultaneously provide an extra margin of safety in case something happens to the primary prerecorded program source in the middle of a performance that is relying on extensive amounts of taped material. (The tape or leader may break, the machine can malfunction, etc.)

Because there are always slight variations in tape speed from one tape machine to the next, the pair of decks will be in as close to perfect sync as possible without the use of timecode. Should one machine get a little ahead or behind, a VCO may be used to re-align

CLICK TRACKS

clear leader acts as a safety precaution to prevent the next song accidently starting before its time. Although the tape machines are located at the house console, Reed is sometimes preoccupied with other aspects of the performance, such as setting up a new mix, when the tape segment comes to an end. With only a couple of inches of clear leader between each song, there is not much margin for error. Reed also writes the name and duration of the following song on the leader so that he is always positive the right portion of the tape is cued. "Once the song starts, the tape better be right or somebody will be very embarrassed," he says.

Probably the most elaborate tape set-up is the one used on **Cher's** tour with TFA/Electrosound. Having developed a variety of talents as a result of her work in television, movies, and live concerts, Cher's show is a composite of the skills she has acquired from each medium. Needless to say, simulating the effects commonly associated with TV or film is not easy in a live environment.

Engineer Andy McCartney was brought in specifically to operate two reel-to-reel Otari MX-5050 four-tracks that contain practically the entire show on tape.

"We recorded the tracks so well in the studio that nine out of ten times the audience doesn't realize that we are using tapes," he says. "If I don't have a separate room where I can set up the recorders, we try to hide the tape machines or at least make them inconspicuous whenever possible, I just make believe I'm a representative from a radio station taping the show."

The track assignments of the four-track master look like this: *Tracks #1 and #3* are the basic music tracks recorded in a studio prior to the tour; *Track #2* is the lead and background vocals and dialog tracks; and *Track #4* is the click track for timing.

The backing vocals on track #2 were recorded in such a way that they act like a click track; the band can keep the beat by the singers. The dialog includes introductions and Cher's raps with the audience, which take place while she is backstage changing her costumes.

As with the other shows discussed here, the clicks on track #4 are fed to the band through headphones, while the music goes through the monitors on stage. Each track on the tape machine has its own channel at both the main and monitor desks.

ACCOMMODATING SONG CHANGES

Cher's original act was composed of almost a dozen costume changes scheduled throughout 30 different musical segments. During the course of the two-hour show, McCartney was responsible for 32 separate tape cues. Fortunately, the show was eventually cut down to 55 minutes. However, to stay up-to-date on the road, McCartney had to keep editing the master tape to conform to the ever-changing format. Whenever a song was cut out, he had to remove that entire section from the tape. And, because each of the eight dancers on stage has a line to sing, or actually mime, during the *Cowboy* portion of the show, McCartney is responsible for recording the vocal tracks every time there is a replacement in personnel.

"When a new dancer with a different name joins the tour, I have to record them with a live mike, usually just before the show, and try to get their voice to blend with the rest of the tape. We use a lot of EQ to get the live sounds to match the prerecorded ones."

In addition, certain changes are to be expected with each new room or opening act. At Caesar's Palace in Las Vegas (where the tape show is augmented by a 60-piece house orchestra) and in similar show rooms, the opening act is usually a comedian. The standard introductions, recorded in a studio prior to the tour, are customized for the particular comic simply by dubbing in the correct name just before showtime. "Ladies and gentlemen, introducing (insert name of act)."

Next comes Cher's taped introduction and the prerecorded opening music, which the band (the drummer, guitarist, percussionist, bass player, and two keyboardists from Ed McMahon's *Starsearch* television show) mimes to as they do for about three-quarters of the songs. (Only two or three tunes are actually performed live.) From then on, much of Cher's material is characteristic of a television variety show, such as an entire routine based on Rod Stewart's "Do You Think I'm Sexy," with Cher dressed up as her character Laverne.

Except for one instance where the machines keep running, McCartney has to repeat the same recuing process after each song, so that the prerecorded material doesn't accidently leak into the house system, and tip off the audience to the use of tape. (The program faders on both mixing consoles — house and monitor — are always left open.) When a song ends, McCartney must turn off both tape machines, set to "off" the toggle switches on the A-1 Audio-built interface boxes, manually wind the tapes to the next cue, cue them up perfectly

OPERATION OF CLICK-TRACK SYSTEMS — continued . . .

the two tapes.

"Since live shows are pretty fast-paced, there is no chance to resplice the tape, or to even re-thread and wind a new tape to the appropriate place," says A-1 Audio's president Al Siniscal. "Using two machines is the best way to keep the show going with a minimum of embarrassment, and hassle. When we throw the toggle switch that causes both machines to enter play mode at the same time, it may not be in perfect synchronization but, for all practical purposes, it's reasonably close. The band may be off by a half a beat momentarily, but when they move on to the next song, everything is back to normal.

Although the console operator is capable of starting the tape machines, A-1 usually sets up remote starts for the conductor. Being on stage and in close contact with the performer, the conductor is better able to determine when the song should begin. The start button is built into a stage panel next to the conductor's piano. Sometimes two or more remote-start locations are available, depending on the conductor's needs. □□□

CLICK-TRACK CASE STUDY: RICK SPRINGFIELD TOUR

Rick Springfield recently completed a tour to support his new album, *Hard to Hold*. Accompanying him on the road is engineer Andy McCartney, and two Otari MX-5050 four-track machines to replay primarily studio-generated sound effects that are too complex to reproduce live.

McCartney, Springfield, and producer Bill Dresher spent several hours in the studio remixing a quarter-inch four-track tour master from the album's original multitrack masters. McCartney then ran off additional safety copies during rehearsals. Channel assignment of taped material is as follows: track #1 and #3, musical sound effects and some music solos; track #2, click-track and countdown vocals; and track #4, background vocals. The engineer refrains from explaining precisely what is recorded as "music" tracks, because "that can give the game away." He does say, however that he is responsible for replaying at least one guitar and several keyboard effects from the album. More traditional types of music parts are also pre-recorded, but as to which they are, the engineer again won't say.

McCartney has approximately nine major tape cues throughout the performance, with some major cue comprising two or three minor cues. For example, one song may contain three cue points, another may have two, and so on. For this and other reasons, McCartney offers that this particular application of a click-track system is somewhat more difficult from previous tours he has worked on. Rather than simply turning the machine on at the beginning of a song and shutting it off when the tune is over, his role is more like that of a studio mixing engineer.

The tape machines start the show off, and continue for the next three songs before McCartney gets a break. The first song runs straight through, while second one requires that he start, stop, wind-forward to the next cue, and then restart the tapes again on a certain beat in the middle of the tune. "There are cues where I have to take one channel out, and mix others in, all at precisely the exact beats," he says. "One song, for example, has three cues, each on separate tracks, occurring at odd intervals, and all for different lengths of time.

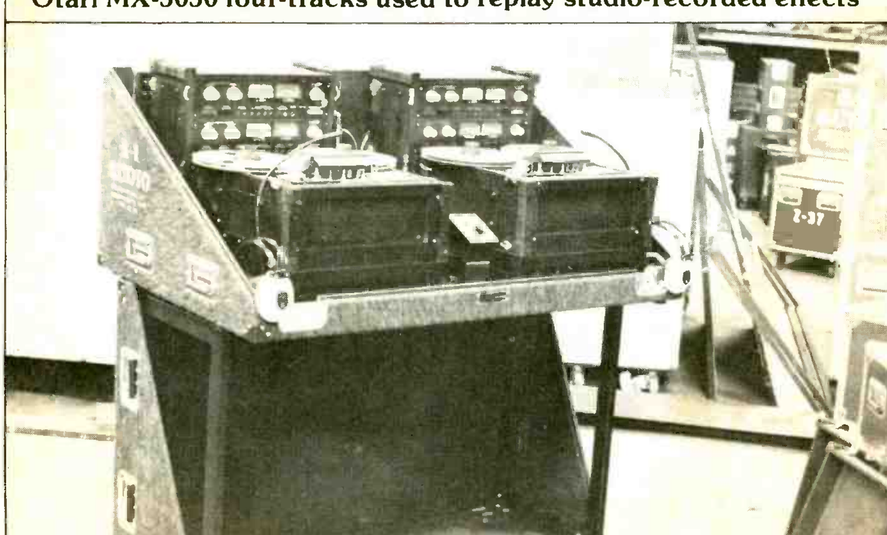
"Rick Springfield commented the other day that I'm really another member of the band, because of all the intricate timing needed to coordinate the show. I take my cues from him, and the rest of the band takes their cues from me. There's a very delicate communication balance that has to be maintained.

"During rehearsals, I was confused about one of the tape cues," he recalls, "and brought the track in at the wrong place. It really threw everybody off. The pressure here is much more intense than with the Cher show."

The mixing is so complex, McCartney continues, that he is called on to alter the pitch of program material in the middle of one song. "Because one of the tracks has to be raised in pitch, and then brought back to normal for the rest of the channels for a later cue, I have to align the heads of both machines with a guitar tuner whle the song is going on. I have to be very careful, because the pitch control is extremely sensitive; the track can go sharp or flat real easy."

As with other systems, the drummer is dependent on the clicks for maintaining strict musical tempo. Yet this stage show also incorporates a video presentation,

Otari MX-5050 four-tracks used to replay studio-recorded effects



CLICK TRACKS

so that the click and music for the next song come in as soon as he hits the start button, reset the toggle switches, and start both machines together at precisely the right moment.

Needless to say, McCartney's physical location is very important. "I'm always out front because I have to take the cues from Cher, the conductor and anyone else on stage who may affect the flow of the show." (An exception to this occured on the HBO taping of Cher's show at Caesar's Palace in Las Vegas. The video cameras were positioned in such a way as to obstruct McCartney's line of sight with the stage. To get a clearer view, he moved himself and the tape machines to the video truck, and took his cues from the video monitors.)

To compound matters, Cher sometimes cuts a song from the regular line-up during the course of a show. In the short time while she's talking with the audience, McCartney must make a mad search for the tune on two reels of tape, in addition to the regular recuing process mentioned previously. On A-1's recommendation, each song is separated from the next by only four inches of leader that contains the name and length of the following song. If the leader was any longer, the few seconds between songs wouldn't give McCartney, or any operator, enough time to line up the next cue.

Because the Otari decks are not equipped with varispeed controls that enable the tape to be varied in the event that one gets a little ahead or behind, McCartney has to vary the speeds by hand. "Very carefully, I put my finger on the reel, and roll it a little bit faster to catch up with the first machine. Or maybe slow down one machine manually with my finger on the reel to match the two up. I have a headset from both machines that I listen to simultaneously along with the house sound, which may be on a delay of some kind. It gets pretty confusing sometimes!"

As if the time delay isn't enough to contend with, power fluctuations also come into play in certain parts of the world. "We often have to use transformers to hold the voltage level constant," says McCartney. "In South Africa, we were actually running two ips slow for all the shows!"

"Doing monitors and the house mix are demanding jobs; I've done both," he continues. "But I think operating the tape machines is tougher. Sometimes the people on stage forget what

Conductor's control center and music stand, with Auratone monitor, BGW amp and Shure mixer.

to do or say, and I have to anticipate that. There are times when they're not in the place where they should be, and everybody ends up standing around on stage waiting for the tape to start. The show is running on my timing. If I get it wrong, the show screws up."

LOW-COST CLICK-TRACK SYSTEMS

Jack Jones expands his show with a Tascam cassette deck positioned next to the conductor's piano. Each song is recorded on a separate cassette, and each tape is cued up before the show. At the appropriate time, the conductor slips the tape in the machine, and the band can monitor the background vocals without really having the singers present. According the Ken Newman, the members of Jones' band say the system is very reliable.

Newman also has seen click systems built around a single reel-to-reel machine but without an external amplification system for headphone cues. The two-track is located next to the conductor, and the headphones fed directly from the machine's headphone jack. Newman warns, however, that this may not be practical for groups that play music with high stage levels. The live music will probably mask the click, and time synchronization will suffer.

So, regardless of what is needed to make an aural impression, tape systems may be the best low-cost answer. With a little planning, practically any sound can be reproduced in alive setting — even applause. ■■■

RICK SPRINGFIELD TOUR — continued . . .

which the drummer and band must accompany even though they cannot see it directly. Again, the synchronization of music with visuals is McCartney's responsibility. The drummer, who the band follows musically, takes his cues from the click-track coming to him through a headphone system. McCartney must start his audio tapes at *precisely* the right moment to align the soundtrack with the video.

McCartney's part in the overall performance is compounded by the fact that he is still operating two Otari four-track machines in "practical" sync. He must maintain synchronization manually throughout the entire performance by cueing two reels of tape for every song cue, and continually monitoring both decks to ensure that their speed stay as close together as possible.

To make McCartney's responsibilites a little easier to bear, a custom rack/road case was built by A-1 Audio, sound supplies for the tour. Lou Mannick, who is in charge of A-1's fabrication department, mounted the pair of four-tracks on a welded steel frame that is shock-mounted within a large shipping case. On the frame above the Otaris are mounted various switching and control electronics, and a couple of work lights. Special heavy-duty rubber shocks with internal steel fittings isolate the tape-machine frame from the case, so that all impacts to the case during travel are absorbed before reaching the delicate electronics and tape-transport mechanisms.

The shipping case has also been designed for ease of everyday use. During setup, the case is simply rolled out of the truck. When set up in position, the cover is removed and the case acts as the stand for the equipment. The case's oversized dimensions allows cables and power wires to remain plugged in during shipping, thus eliminating the need to repatch before every show.

The Otari MX-5050 four-tracks have been heavily modified. Installation of special sensing posts ensure that the recorders turn off automatically between cues, which acts as a safe-guard just in case the operator cannot get to the "off" switch before the following cue hits the playback head. Remote connectors have been added so that the conductor or someone other than the tape-op can start and stop the machines.

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SYNTHESIZERS IN THE STUDIO

RECORDING ORIGINAL MUSIC AND SOUND EFFECTS FOR A VIDEODISK PROJECT

MIDI-Controlled Sequencers and Synthesizers Linked to Sync Tracks and SMPTE Timecode for Audio-Visual Sweetening

by Tim Bomba

Today, with large numbers of film and video projects being produced for a variety of industries, more and more time is being spent on providing each project with as much identity as possible. Original music scores and effects have become one such way to highlight the visual aspects of a multimedia project. Recently, this writer was contacted to write and produce original music and effects for an AT&T VideoDisk to be used at the 1984 Summer Olympics in Los Angeles.

The project turned out to be a 10-minute presentation describing the AT&T Unix computer system; the video contained two computer-animated pieces, one of which ran one minute, and the other 1½ minutes. The executive producers and designers of the animation, Belove-Laiserin, of Princeton, New Jersey, had created the animated pieces on a budget that didn't allow use of the more powerful, and expensive, computer systems normally used to create visuals on multi-effect films. Instead, the animation was created on a Dubner CBG-2 computer which, although restricted because of the limited amount of data it can handle, produced some very creative procedures (colors, shadow perspectives, etc.).

At this point, the normal procedure would be to add library music and some standard sound effects. To give each of the pieces character, however, the executive producers wanted an original score with effects. Since one of the animated pieces featured 72 visual cues in 90 seconds, laying down an assortment of effects in a film-sound studio would have been cost prohibitive. What was needed, obviously, was a method of accommodating the project, while still keeping within the budget.

Since large orchestration was completely out of the picture (if you'll excuse the pun), the obvious solution was to perform the entire piece on synthesizers. Because of his previous experience with MIDI-based systems, I contacted keyboardist/co-writer Bob Kinkel to work with me on the AT&T presentation. The combination of a custom computer program and MIDI interfacing of a sequencer and several synthesizers has resulted in the development of a cost-effective system that: 1) allows the majority of work to be done outside of a full-service recording or film-sound studio; 2) allows complete scoring and sound effects synchronization to picture; and 3) makes more productive use of studio time.

Synchronization to Picture

The first step in the process was to obtain final edited copies of the video material with SMPTE timecode burned into the picture. (Both half- and ¾-inch videocassette copies were needed, since we had a VHS deck to work with at my personal studio, and a U-Matic once we entered the recording studio.) Having timecode burned into the picture allows the start and stop marks of all effects and changes of scenes to be accurately computed, and was necessary to show us where to begin and end a particular sound effect for a specific visual effect. To do this accurately necessitates the use of a videodeck equipped with freeze-frame and single-frame advance options, since some effects lasted only five video frames, or 167 milliseconds, and would have been impossible to spot at regular viewing speed.

As is standard practice, timecode had also been recorded onto one of the videocassette's audio tracks, to enable subsequent sound-effects and music tapes to be synchronized with the picture for layback to the master videotape reel.

Music Development

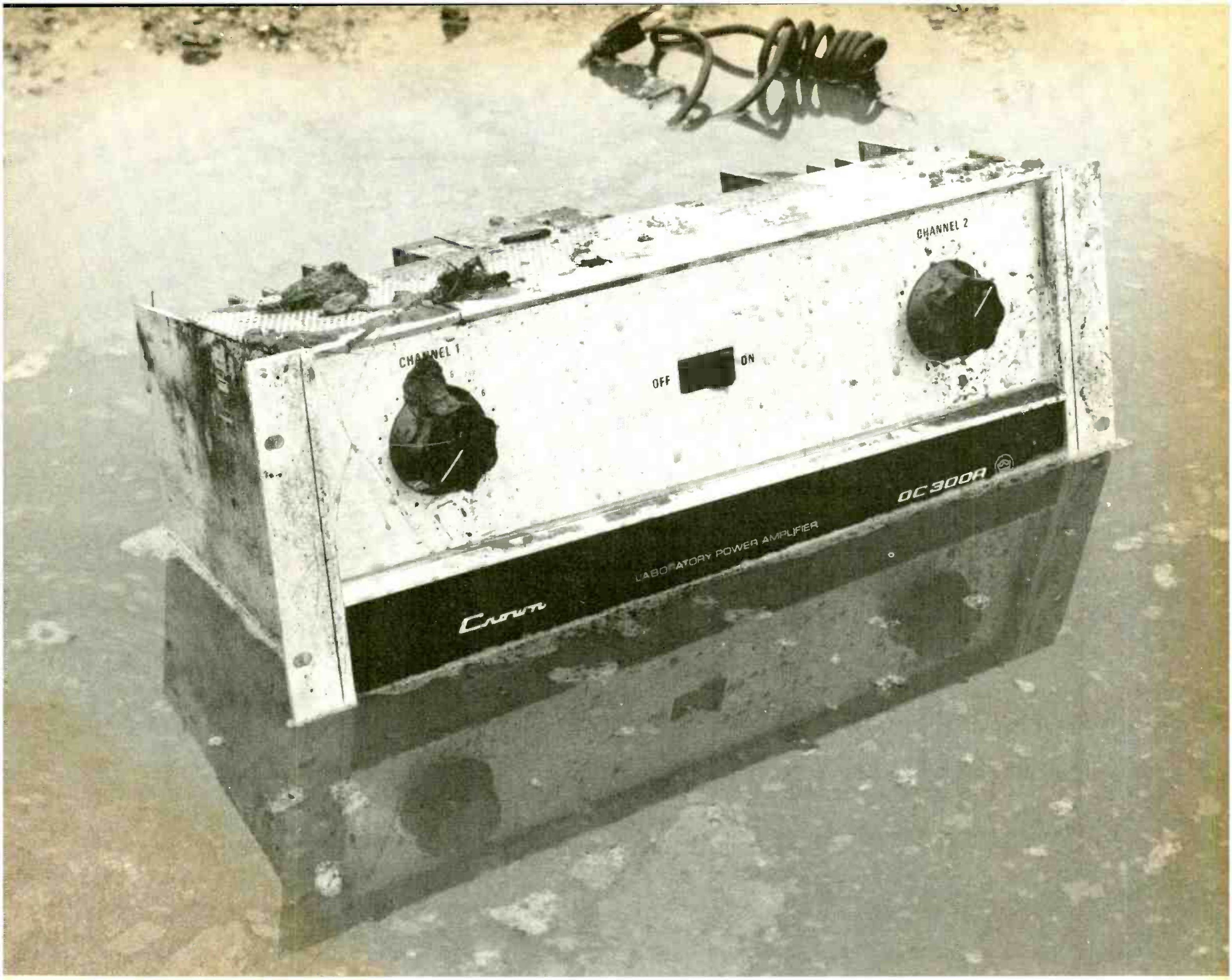
Next came the development of music themes and tempos, which was done mostly by feel. Since the first piece showed an animated robot figure, the accompanying music was written to complement a "technical" feel. Also, because this piece contained the previously mentioned 72 visual cues in 1½ minutes, all of which required corresponding sound effects, the music had to stay simple and not interfere with the multiple effects. And, with all this movement, the tempo had to be up so that it matched movement on the screen. Since the second piece was a more dramatic idea of a city coming into view via a slow zoom in with fewer visual effects, the music was considered more important, and the sound effects less overt. (Some effects were even written into the music as chord changes.) As it turned out, even though the tempo was about the same for both pieces, the feel was quite different for each.

Once the general mood for each piece had been decided upon, various musical ideas came to mind. There were a few compositions Bob and I had written previously and which, with a few changes and timing adjustments, fitted the video pieces very well. Now that a musical identity had been created for each piece, we needed a method to start the effects and properly synchronize them with the music and video sequences.

It was now that the custom program for our Commodore 64 came into use, and which we initialized by feeding in the tempo and time signature of a particular piece. Having typed in each of the SMPTE timecode locations that corresponded to a visual cue, the program told us where each of

— The Author —

Tim Bomba is a producer/writer/engineer that works primarily out of Record Plant Studios, New York. He has a strong background dealing with timecode in the studio, and worked for six months on the MUSE *No Nukes* project, overdubbing and mixing tracks. He also worked on the Queen movie.


CHANNEL 1
OFF
ON
CHANNEL 2
DC300A
LABORATORY POWER AMPLIFIER
Crown

the code numbers were to be placed in the music score, accurate to a part of a beat and the nearest 32nd note. Our final "score" for the effects would print out something like the example given in Table 1.

Additional features of the program include: the ability to change tempo or time signature simply by feeding in the updated information and generating a new printout showing where the cues now fall; working with drop (59.94 Hz) and non-drop-frame (60 Hz) timecode, as well as converting film footage and frames or 24 f.p.s. timecode; an option to enable the computer to calculate a tempo for the composition that will maximize the amount of visual effects falling on the beat and which saves time on the sequencer, since you're dealing more with downbeats as opposed to 32nd notes of the beat — if the user finds a tempo that feels good for the music, the program then will find one close that also fits the visual effects; and storage of up to 500 cue points.

The reason for a 32nd-note resolution on effect timing was that we were using a Roland MSQ-700 Sequencer to store this information, and the MSQ-700 works down to that interval. At the tempo being used, this resolution meant that our effects would be accurate to within two frames of video, or 67 milliseconds. After the entire "score" was printed out, we were ready to begin music composition.

Synthesist and songwriter Bob Kinkel (left) with author Tim Bomba in the latter's personal-use studio during scoring of music and effects for the AT&T videodisk project.

MIDI-Controlled Sequencer

The writing process centered around a Roland MSQ-700 Sequencer, into which was loaded information via a MIDI hookup connecting the sequencer and various synthesizers, including a Sequential Circuits Prophet Five (retrofitted to handle MIDI), Yamaha DX-7, SCI Drumtracks drum machine, and Minimoog. Next came the main theme recording into track #1 of the sequencer. (Reference to different sequencer tracks means digital data storage rather than analog tracks.) When this stage was completed, the main theme served as a guide for effects recording.

The sequencer was now running on its own internal sync, and could play back the main theme automatically. We had a half-inch VHS videocassette deck to view the animation against music, and constantly double check that we were in the ballpark. The videocassette would be started, counted down with the burned-in timecode showing on the video monitor, and the sequencer started by hand at the beginning of picture to play back the theme. [Currently, the MQS-700 sequencer cannot be triggered by timecode — see accompanying sidebar for details of recently announced SMPTE-controllable sequencers — *Editor.*] Triggering the sequencer by hand on a visual cue from the video wasn't the most accurate technique, but it gave us a good idea if we were close to the visual timings. (Later in this article we'll consider correcting for human error.)

After the theme was recorded and in place, we next moved on to the effects. One important point to be made here is that indiscriminate sound would not have worked for each *visual* effect, since the sound effect had to both complement the visual effect *and* work with the musical theme. If an effect sounded too musical, it might mask or be masked by the main theme. And we were still attempting to give each effect its own identity.

Going back to the computer program, the start and stop point of each effect was now translated into musical terms — measures, beats, etc. — and in terms the sequencer would understand. Let's say that the first effect started in measure #2, beat two and on the third, 32nd note of that beat, and last for five frames. After deciding what the effect would sound like, and on which synthesizer it was to be created, we would go to the sequencer, find the next available data track (channel), and put it into record-ready mode.

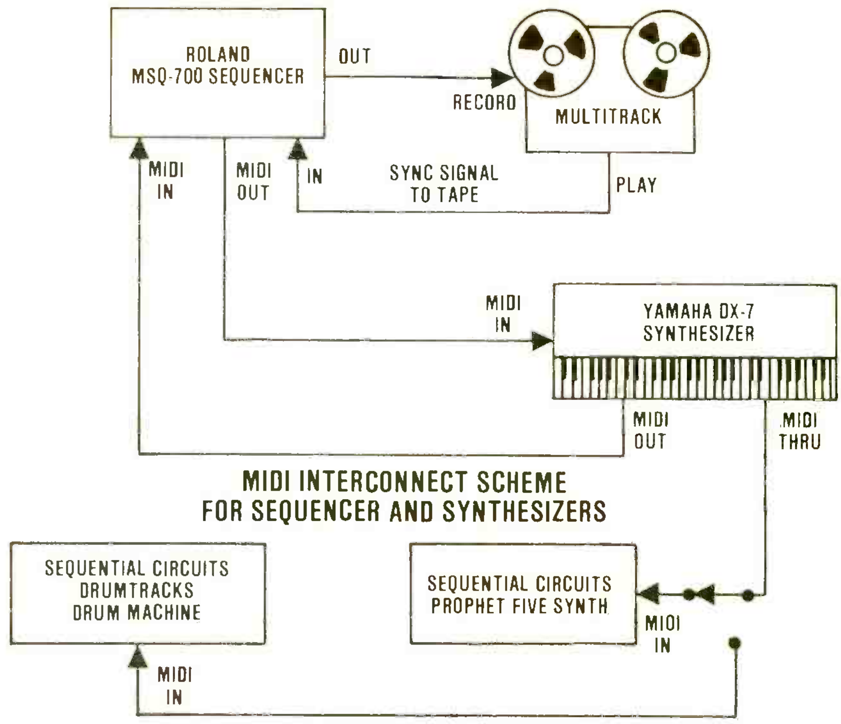
Because the first effect doesn't start until a portion of the second beat of measure #2, we programmed rests into the channel until the point at which the effect begins, including the two 32nd-beat rests that lead up to the

Effect Name	SMPTE	Measure	Beat	32nd note
Robot Arm Up	00:00:01:24	34	2	3
Robot Arm Stop	00:00:01:27	34	2	7

TABLE 1: Sample output from Custom Program to calculate musical notations corresponding to SMPTE timecode locations.

EQUIPMENT USED IN PERSONAL-USE STUDIO DURING AT&T RECORDING PROJECT

- TEAC Model 244 Portastudio combined 4/2 mixer and four-track cassette recorder.
- TEAC A-3340 quarter-inch four-track.
- MIDI-retrofitted Sequential Circuits Prophet Five.
- Yamaha DX-7.
- Sequential Circuits Drumtracks drum machine.
- Roland MSQ-700 MIDI Sequencer, with Sony Pro Walkman for data recording.
- Yamaha PM-170 mixer.
- MiniMoog
- DeltaLab Effectron ADM-1024 and ADM-64 digital effects processors.
- Ibanez GEI-502 graphic equalizer.
- dbx Model 160x limiter.
- Rockman X-100 (Put your synth through this gadget — it creates some nice guitar sounds, even with the synth. And in the distortion mode the X-100 is quieter than its predecessor — TB)
- Commodore 64 computer running custom software (see main article).



MIDI INTERCONNECT SCHEME FOR SEQUENCER AND SYNTHESIZERS

effect start on the third 32nd beat. At the beginning of the effect (the third 32nd note), we played a key on the Yamaha DX-7 to indicate to the sequencer that a signal would now begin, and tied the duration of the note for five frame. Rests were then programmed into the sequencer for the duration of the track — unless, of course, this particular effect happened again in this piece, in which case we repeated the above procedure at the required point. The next effect would go onto the next available data track (one per effect), and so on until the sequencer's eight tracks were filled up. (Although bouncing and combining of data tracks on the MSQ-700 is possible, for our purposes working with one track per effect provided best results.)

Because the Roland sequencer records only the effect cues (start, stop and duration) and not the particular sounds themselves (for example, white noise, violin sound, and so on), a list was kept of what sound corresponded to each effect. When all eight data tracks were filled, the sequencer data was saved to cassette via a Sony Pro Walkman. Sequencer memory could now be erased, the procedure repeated for the next eight effects, and so on until all effect cues were recorded. As yet, no audio recording to tape had taken place, although sequencer data for the entire composition with effects (in sync) was complete.

To test out the procedure, we reloaded the sequencer data from cassette into the MSQ-700, eight tracks at a time. Then we recorded a rough mix of the entire piece onto the TEAC Model 244 Portastudio, playing back one data track at a time to run individual synthesizers. Since, in addition to the main them, there were some 30 different sound effects, we had to keep bouncing down audio tracks on the Portastudio. Also, because we didn't have SMPTE lockup at my personal-use studio, the rough mix, when run with the video, wouldn't be in perfect sync. However, by simultaneously starting the videodeck with the Portastudio, we could get close enough to see the visual and sound effects line up. Now a rough mix of the compostion could be played to the executive producer (who at times was difficult to reach) via a phone patch to his answering machine, to let him hear how the project was progressing.

Trial mixes were laid onto a TEAC Tascam Model 244 Portastudio, seen here with a SCI Drumtracks MIDI-controlled drum machine.

Tracking Music and Effects

At this point, we were ready for the recording sessions at Roxy Recorders in New York. (We chose to track at Roxy mainly because I couldn't book time at the Record Plant.) Normally in such circumstances, we would have opted for a servo-controlled multitrack that can be slaved via a SMPTE synchronization system to a videocassette deck replaying the visual material. In our case, however, we had decided to work with an Ampex MM-1000 16-track that doesn't feature servo-control. Because we were under budget pressures, and decided to use the MM-1000 — which we also knew would provide excellent audio recording quality —some method had to be devised to slave the videodeck to the multitrack. As it turned out, we rented a customized video system from Audioforce, the New York-based hire company, and got more than we bargained for.

Audioforce president Sid Zimet has developed a special interface between a modified JVC ¾-inch videocassette deck and a BTX Shadow synchronizer that allows the U-Matic to *slave* to any timecode source, rather than act as a master. In our case, the ability to have the videodeck follow to the multitrack enabled the recording process to work as it normally would, without worrying about video synchronization. For example, while overdubbing to tape, the multitrack normally would be running as a slave to the videodeck. With about seven seconds needed to lock together the two machines, there is a chance that the multitrack would not be running at the correct speed during its lock-up time. If the overdub went down while the multitrack's speed was still locking up to the video machine, the subsequent playback would possibly have some wow on it. With the video acting as slave to the multitrack, the engineer is assured that the audio tape is always running on-speed.

The first step was to lock together the videodeck and the multitrack. To do this, SMPTE timecode was taken from the videocassette, put through a timecode reshaper, and transferred to track #16.

Next we went to the MSQ-700 sequencer and set our tempo for the first animated piece. We started the multitrack about 20 seconds prior to where the first animation would start, which offset allowed the videodeck to catch up and synchronize with the multitrack. (Normally, seven seconds is plenty of offset.)

Counting down to the start of the picture by watching the SMPTE timecode display in the bottom of the video picture, we started the sequencer for its first run at approximately the beginning of picture. This first run was used to print sequencer sync pulses onto the multitrack roughly in time with the timecode. (Sequencer sync was recorded on track #14 to ensure no cross-talk between it and timecode.) But, because the sequencer had been started by hand, and the start point didn't exactly correspond with the picture, we had to offset the video a little to accurately follow the sequencer sync pulses recorded on the multitrack. Since we could see that the sequencer sync was behind — we had been late while hitting on the downbeat — all effects now being run automatically by the sequencer would be behind the video. (Obviously, the only way to tell exactly how far off we were was to print one effect onto the multitrack and play it back against the video.)

Going to the BTX Shadow, we programmed in a retard offset for the videodeck. For our purposes, the videodeck had to be moved back just seven frames; running the multitrack and the video showed that this indeed was the correction factor necessary to bring the sound effect into exact sync with the video. Calculating the required offset by trial and error took about five minutes.

Now that everything was running

A Roland MSQ-700 MIDI-compatible Sequencer provided sync pulses during the music and effects recording sequences.

in sync via the offset, we proceeded with audio recording of the main theme and sound effects. We would reload the data into the sequencer cassette file by cassette file — each file held up to eight tracks of information — put date track #1 into playback mode, lock up to whatever synthesizer sound corresponding to that track, program the sound on the synth, and run the multitrack, which, in turn, was now controlling the sequencer via the sync pulses recorded on audio track #14.

The videodeck locked up to speed, and we watched as the visual cue came up; the sequencer triggered the

The author's keyboard array includes a Minimoog, Yamaha DX-7, SCI Prophet Five, and Fender Rhodes.

synthesizers; the synthesizer sound was recorded onto its own audio track; and everything — audio and video — was perfectly in sync. This procedure was repeated for all eight data tracks on the sequencer, after which a new file of eight data tracks would be loaded, and so on until the entire piece was recorded.

Total recording time for music and about 55 effects needed for a 90-second composition: two hours. Normally to sync up effects to this many visual cues in a recording or film-sound studio could require between two and three days; at $200 to $300 per hour the process can become very expensive.

Remixing the multitrack tapes was pretty basic, since the final two-track mix would contain a mono mix of the theme and effects on track #1 and SMPTE timecode on track #2. In addition, the same mix was laid onto an open track on the multitrack in Sel-Sync mode, to provide a safety copy.

Needless to say, the final film-sound mix session at Reeves Teletape, New York, went just as quickly. We put up the music/effects mix, hooked up the SMPTE track to the facility's synchronizing system, and were ready. However, we had to remember that, because an offset was needed between the video and audio timecode during the recording process, and the same SMPTE code was used throughout the entire project, master code was now on the video and all audio code on the slave. We therefore offset the audio back by the same amount of frames as the video had been advanced earlier, ran the video with our mixed track, and it lined up perfectly. This procedure took about 10 minutes, and saved a small fortune in studio costs.

Future Developments

The advantages to such a system are clear. With the high cost of today's video post-production facilities, a sequencer-based system like this can help to complete a project more efficiently. And, with the amount of productions in progress, and even in planning stages, the cost effectiveness becomes a significant factor. Since this first project, and during subsequent projects, there has been time to fine tune the procedure, and even come up with some ideas for the equipment manufacturers.

Since the Roland MSQ-700 sequencer formed the heart of the effects procedure and, combined with Bob Kinkel's computer program, saved a lot of studio time, there are a few modifications that have taken place, and a few that would be nice to see in future products. If you are laying in five similar effects on a sequencer data track, and the third isn't in the proper spot, but the fourth and fifth are fine, you can punch in on the data track just before the third and re-align it. However, because there is no punch-*out* facility, you have to rerecord completely to the end, including effects four and five.

According to Dan Desouza of Roland, the new MSQ-100 Sequencer edits, copies measures, and performs a number of useful functions. In this way you can take a track from the 700 with multiple MIDI channels, bounce it over to the 100, edit the data, and bounce back to the 700.

Another useful feature would be the ability of a sequencer to record synthesizer control settings for various effects; currently, we had to write down each synth setting corresponding to sequencer data tracks. Also extremely handy would be a way of deriving MIDI clock data directly from SMPTE timecode, which would allow automatic triggering of the sequencer at the start of a code frame. And, if a continuous piece of music is needed for more than four minutes, a sequencer sync pulse train sometimes will drift slightly, which means restricting the piece to a two- or three-minute duration.

Roland's new SBX-80 Sync Box, a device that generates MIDI clock from the SMPTE timecode recorded on tape, ensures that the MIDI clock running a sequencer will be accurate for theoretically any length of time. And, you won't have to print a sequencer sync track on the multitrack tape.

Since, in this writer's experience, the Yamaha DX-7 is one of the most popular synthesizers in use today, it would be an advantage if the unit could send MIDI information on *all* 16 MIDI channels, instead of just channel #1, allowing users to take advantage of the Roland sequencer's built-in multitrack and bounce-down facility. (The DX-7 does receive on all 16 MIDI channels, however.) As for the Sequential Circuits Prophet Five, in order to record more than one effect per sequencer track, it is necessary to retrofit the Prophet with the Poly-MIDI System. Such modifications, especially those from Roland, seem to be happening at a fairly respectable rate in response to consumer suggestions. ■■■

INTERFACING MIDI-EQUIPPED SYNTHESIZERS AND SEQUENCERS TO SMPTE TIMECODE AND PERSONAL COMPUTERS

A Look at Some of the Interesting MIDI Hardware Unveiled at the June NAMM Show

by Bobby Nathan, Unique Recording Studios, New York

Many studio engineers and producers have experienced what MIDI is all about at present, but some of the new products shown at June's NAMM Convention in Chicago will soon take MIDI a step further.

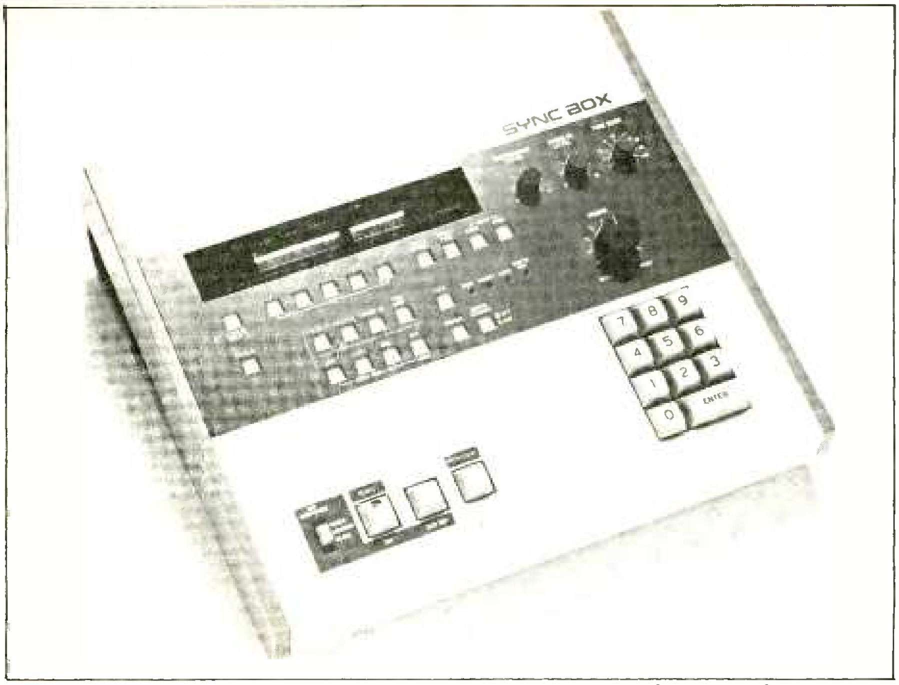
The most important letter in the word MIDI is the "I" for Interface, and J.L. Cooper Electronics' selection of interface devices can only be described as staggering. The Channelizer can give any synthesizer MIDI-send channel capability. Many of the home-computer MIDI sequencers need a defined MIDI-Send channel to enable that track to play back on a specific MIDI channel. Cooper's Channel Filter gives synthesizers the added capability to receive on a particular channel (i.e. synthesizers without MIDI-Receive channels). The MIDI-to-CV Out and In boxes can drive, for example, a Minimoog from a Yamaha DX-7, and vice-versa. The Drumslave and Braindriver, respectively, will interface Simmons drum pads to a MIDI sequencer, and then the MIDI sequencer to a Simmons head.

Roland's MM-4 MIDI-Thru box is very low priced, and offers one MIDI input and four MIDI outputs. It definitely cuts down the timing delay that results from patching one instrument's MIDI-Thru jack into another instrument's MIDI-In jack, etc., etc. Also, many instruments don't offer MIDI-Thru outputs, so the MM-4 is the only way to link together multiple instruments.

The Korg KMS-30 MIDI Synchronizer is another low-priced interface that is divided into three sections: Tape, Sync and MIDI. The Tape section contains a sync-to-tape output and input. There is a master clock switch that determines whether the sync-to-tape tone is generated in relation to tempo of the sync/clock or the MIDI clock input, a feature that offers sync-to-tape capabilities for drum machines and sequencers lacking this option. (For example: Korg Poly 800's sequencer, Roland TR808, Oberheim DX, Yamaha RX-15, RX-11, SCI's Prophet 600 and T-8.) The Sync section contains a sync/clock input and two independent sync/clock outputs via five-pin DIN connectors (Roland- and Korg-type with 5-volt start-stop pulse). The sync /clock input is selectable between 24 and 48 BPQ (beats per quarter note), while two independent sync/clock outputs are each selectable between 24 and 48 BPQ. The MIDI section contains one MIDI input and two MIDI outputs, and can also be used independently as a MIDI-Thru box, much like the Roland MM-4. The MIDI outputs both transmit MIDI clock in relation to the tempo of the MIDI input or sync/clock input. In summary, the KMS-30 can synchronize MIDI clock devices with 24 and 48 BPQ clocks, and vice-versa, to and from tape.

Roland's new SBX-80 Sync Box can generate and read 4800-baud SMPTE timecode, and can also read either a click track, clock or sync tone from tape, and then output either a metronome, clock, or MIDI clock at all the popular values (i.e. ¼,⅛,1/16 notes, and 12,24,48,72,96,120 beats per minute). The only catch is that it can only output *one* metronome or clock value at a time, unlike Garfield Electronics Dr. Click, which can output many different clocks.

The Sync Box can be programmed to read both SPMTE and the clock simul-



Roland's new SBX-80 Sync Box is capable of generating and reading SMPTE timecode, and can also read a click-track, clock or sync tone from tape.

taneously, thus enabling the user to store cue points in its memory regarding where within the tune to start and stop a drum machine exactly in relation to the tune. If you were trying to sync a drum machine or sequencer to a live drummer, for example, there is a tap button that produces a ¼-note click (to be printed on a track) for the sync box to read on the next pass, enabling it to output the clock of your choice.

The most impressive capability of the SBX-80 was demonstrated at the NAMM show in a hook up with a Tascam 80-8 eight-track. Two tracks were used for the SMPTE and clock pulses, the other six tracks being used for vocals, bass and guitars. The drums were produced by a Roland TR909, and all the synthesizers were being controlled by a Roland MSQ-700 sequencer. Whenever the tape was put in fast forward or rewind and dropped into play, the TR909 and all the sequenced synthesizers waited one bar and instantaneously went into sync at exactly the corresponding part of the song.

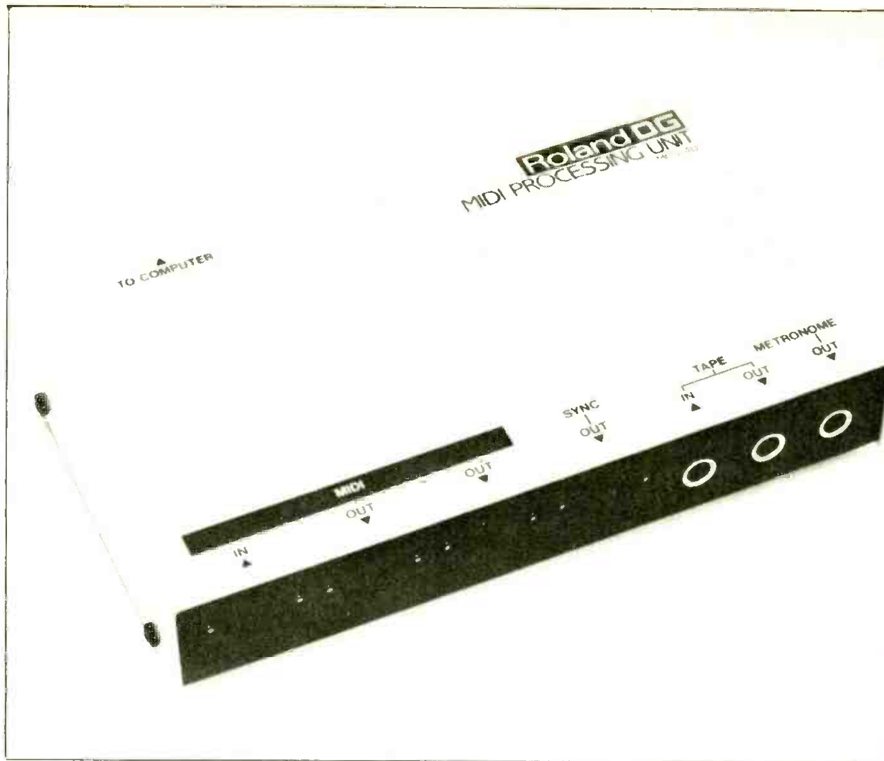
This feat was achieved through the MIDI clock's unique ability to correlate start and end time of the sequence/song to tempo of the clock. Newer machines equipped with MIDI clock — for now, the TR909, MSQ-700, MPU-401, Drummtracks, T-8, Yamaha's RX-11 and RX-15, and the Korg Poly 800's sequencer — all have the ability to know when operated in the Song Mode what bar to go to by reading the MIDI clock.

The state of affairs in sequencers is that mostly all of the add-on types of interfaces for home computers are not as versatile as the stand-alone units, including the Oberheim DSX, MSQ-401 and Yamaha's soon to be released QX-1. The exception is Music Data's sequencer for the Commodore 64, which can record pitch-bend, velocity and modulation information. The unit features 16 tracks per sequence, 16 sequences total, and 64 steps to the song mode. All 64 steps of the song are numbered on the screen display, and the sequences that you have chosen are next to the appropriate step numbers. Also, the 16 tracks of the sequence you are currently working on are displayed simultaneously. In addition, the MIDI channel number for each track and the play status (mute) are shown clearly. The 10,000-note sequences in memory can be stored on the Commodore's disk drive, and can even be sent over the phone via a modem.

In this way a songwriter in L.A. conceivably could write a song with another writer in New York.

To me the ultimate stand-along sequencer could very well be Yamaha's QX-1, scheduled for release next year. Currently, the unit offers 32 sequences, eight tracks each, and an 80,000-note memory, although these specs may change. Individual sequences can be assembled into songs, very much like Yamaha's DX-11 and RX-15 drum machines. Sequences are stored on a 5¼ inch floppy disk drive built into the side of this self-contained unit. The device doesn't require a synthesizer with MIDI-Send channel capability for it to play back different tracks on different MIDI channels.

Yamaha's Personal Composer for the IBM PC is a particularly impressive piece of software, because of its music-scoring capabilities on an Epson and other dot-matrix printers. The unit features 32 tracks, with track mute; MIDI channel assignment to the individual tracks; after the fact quantization values; and copy and append functions by bar numbers of individual tracks and sequences. (Imagine a guitar player with a Roland GR700 guitar synthesizer with MIDI writing melodies into the Personal Composer in concert pitch, and then printing out those melodies in the key of E flat for a horn player to read!)

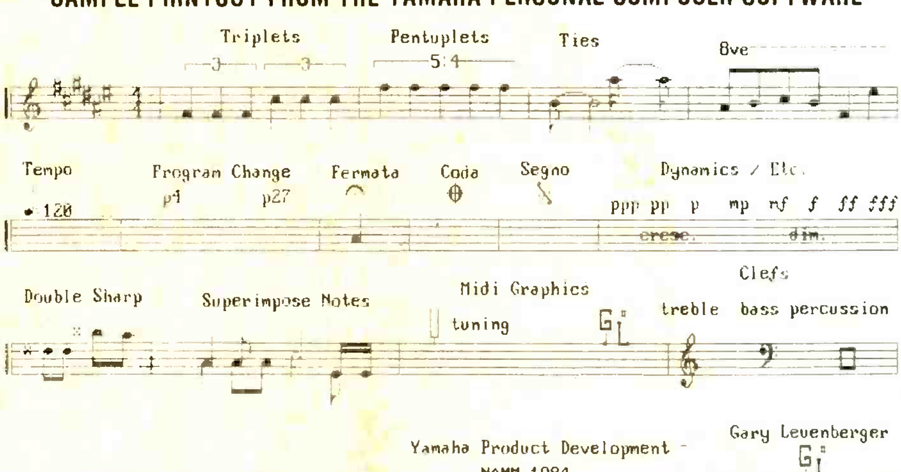


Roland's MPU-401 Interface Controller enables an IBM PC or Apple II to link MIDI-equipped synthesizers, sequencers and drum machines.

Roland's MPU-401 interface controller, available in two versions for both the IBM PC and the Apple IIe home computer, is based on softward developed by Ralph Dyck. (The MIDI control box is identical for the IBM and Apple; the only difference being the plug-in interface card and disk software.) It is set up as a eight-track sequencer with record enable, solo, play on/off, MIDI channel assign, after the fact quantization, one or two bar count off options, MIDI clock, as well as Roland's two new clocks (72 and 120 BPQ). The unit's one-page menu design make it fast and easy to use. Function keys are set up to toggle the metronome on/off, the tempo, and whether internal or external clock is used. There is even a MIDI Echo on/off toggle to turn off the MIDI-Out on the MPU-401 while recording tracks. The MPU-401 box has a metronome-out, Sync-to-Tape out/in, as well as the standard Roland 24-clock, female DIN connector. The only disadvantage to use of the MPU-401 is that it does not offer Song Mode to assemble sequences into a song.

Isn't it a shame that Yamaha and Roland, with their Personal Composer and MPU-401, have not taken full advantage of the IBM's memory expansion capabilities? Hopefully there will be software updates to remedy this in the near future. And

SAMPLE PRINTOUT FROM THE YAMAHA PERSONAL COMPOSER SOFTWARE



where's the interface and software development for Apple's 32-bit Macintosh? And last, but not least, Oberheim's DSX sequencer is my overall favorite with the OB-8 with MIDI. Its only drawback is that the device has to be programmed by the OB-8, but it can play back to any synthesizer equipped with MIDI-In port, especially Oberheim's new Expander, the XP-1.

In closing, the stand-alone sequencers are still favorite amongst producers, engineers, and musicians, because of their built in features, compact size and fast set-up.

— the Author —

Bobby Nathan is a co-owner with partner Joanne Georgio of Unique Recording Studios, New York City. The facility was set up five years ago as a rehearsal studio, and since have expanded into two 48-track rooms with full audio/video capabilities. The studio prides itself on offering just about every synthesizer, drum machine, sequencer, and interface device known to man, as well as almost every piece of outboard gear made. The entire staff of engineers and assistants also are described as being highly skilled in programming. Nathan's "user-application" insights, he tells us, have aided such companies as EMU Systems, Oberheim, Linn Electronics, Garfield Electronics, and PPG Europa in product development.

EQUIPMENT ASSESSMENT

OTARI MTR-12-H TWO-TRACK HALF-INCH TAPE MACHINE

Reviewed by Peter Butt

The Otari MTR-12 is a compact tape machine supplied in either a discrete component configuration, for mounting in a standard EIA rack, or as a console unit with level controls, metering, and channel switching mounted either on a bridge above the horizontal deck plate, or in a canted position directly below and in front of the deck plate. The MTR-12 is a very similar in appearance to the MTR-10, and electronically is nearly identical. The MTR-12 is finished in a light beige enamel, in keeping with the trend toward lighter colors in studio decor and away from the dark browns, blacks and greys that were stylish a few years ago.

Signal interface, AC power, and remote control ports are located on the console's lower rear panels, adjacent to the power supply and motor-drive amplifier heat sinks. All other major electronics assemblies are contained on a series of circuit boards that are inserted into a card cage from the front of the console. These cards pertain to the Record/Play/Bias/Erase systems for each audio channel; Audio Control logic; Master CPU; and Transport Control logic. DC regulators cards are installed in the far-right position of the card cage, while the regulators themselves reside on the rear panel with the motor drive amplifiers, and are accessible by removal of the rear-panel assemblies.

The unit is equipped with an electronic tape counter that indicates elapsed tape length in equivalent hours, minutes and seconds for the speed selected. The timer incorporates a zero-seeking, single-point cue locator system that indicates negative as well as positive times. The tape-time indicator doubles as a direct tape speed readout for fixed and variable tape speeds, indicating to hundredths of inches per second or as a percentage of nominal speed.

The MTR Series are supplied in a choice of three-speed configurations: 3.75/7.5/15 or 7.5/15/30 ips, the high-speed version being evaluated here. Record equalizations may be chosen as either IEC or NAB standard, by means of a slide switch on the Audio Control card. The 30 ips equalization is, of course, always in conformance with the AES proposal. Reproducer equalizations controls have sufficient range of adjustment to accommodate any of the record equalization characteristics provided, and can be compensated for with margin to spare. A low-frequency shelving characteristic controlled by the individual channel reproducer LF equalization control may be selected if unusually wide low-frequency response compensation is required. (This feature was utilized in the test results presented here.) As has been Otari's preference, only a single LF reproduce equalization control is provided for all three speeds, while each of the three speeds has its own separate HF control.

The MTR-12 is a close relation to the MTR-10, employing similar tape path geometry, and a capstan/pinch-roller tape drive system. Apparently, the only notable difference between the two models is the 12-inch reel capacity of the MTR-12.

Facilities are provided for headphone and speaker monitoring of the machines' audio channel output signals. These functions are located in the VU meter panel, along with the input and reproduce level controls, function controls for each channel, and controls for the internal signal generator. The status of each of the audio channels — Record Ready, Record, Input, Sel-Rep, and Repro functions — are indicated by multicolor LEDs. Each of the illuminated VU meters has an LED peak indicator whose illumination threshold is adjustable. The convenience of an editing block is provided.

The MTR-10 was supplied in the increasingly popular two-track, half-inch configuration, a format that was briefly introduced by Studer sometime around the mid-Sixties. It was not then appreciated for its potential, and was abandoned soon after introduction. In early 1978, Ampex quietly demonstrated a two-track, half-inch head assembly for its then newly-introduced ATR-100 Series of tape machines. I was given the opportunity to play with one of those experimental assemblies in May of that year, and found that I liked the wider tape track format almost immediately.

The wide-track stereo format did not begin to attract general attention until after its adoption by Mobile Fidelity Records in mid-1980, using Studer A-80 decks with electronics supplied by John Curl. About a year later, Ampex decided to add the new format to its ATR-100 option list and, as acceptance of the new format spread, Studer, MCI, and Otari were to follow. Half-inch track width is now the format of preference for projects where the highest quality reproduction and dynamic range are desired without resort to noise-reduction devices. It is, therefore, with considerable anticipation that examination of the machine has been undertaken.

Preparation

After unpacking, the machine was powered and spot checked for obvious fault and damage. No need was seen to perform any serious adjustments, save for the routine record/reproduce channel alignments and some ineffective playing with the reel and capstan servo adjustments due to the high DIN-weighted flutter readings noted in the Table 1. It has been my experience that most tape transports will respond to careful, sensitively administered adjustments, and can usually be brought to a performance level significantly better than the values specified for them. This does not seem to be the case for the MTR-12, however; reel motor and capstan servo adjustments do not seem to

INTRODUCING

THE TIME MACHINES
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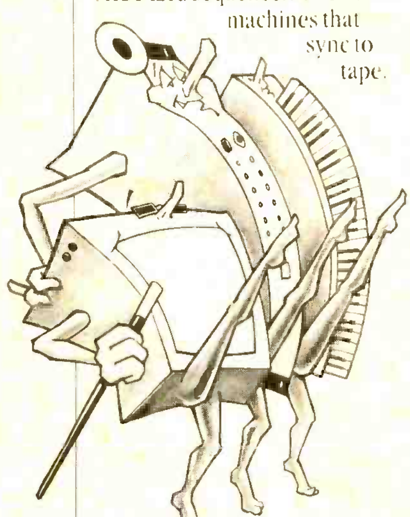

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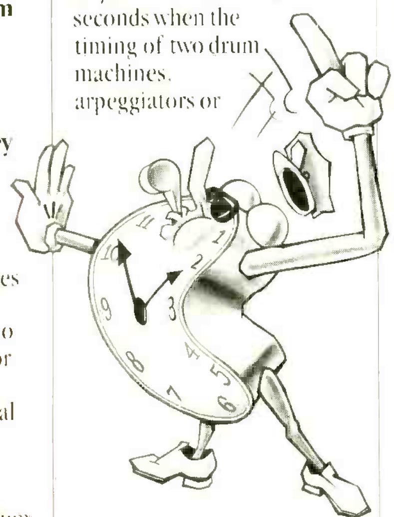

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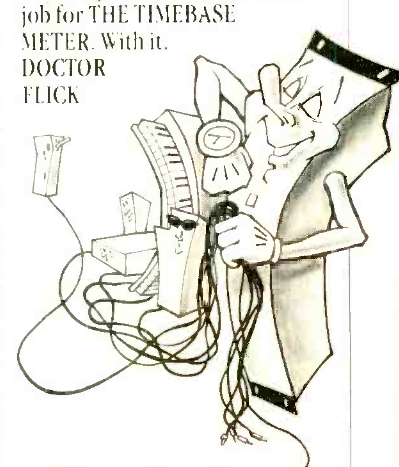

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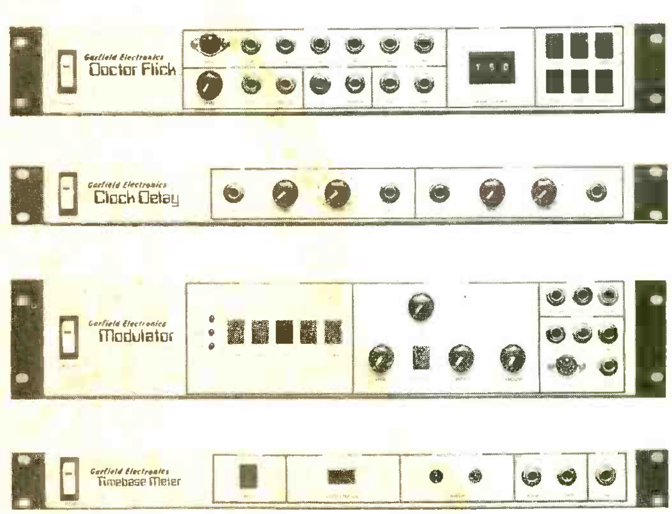

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have any great impact on wow-and-flutter performance within their ranges.

The tape tension control system derives its input from optical reel-motor tachometers without any reference to actual tape tension. Reel-motor tachometer pulses are operated on, based on optically-detected reel flange size, by the main CPU, which also controls all deck functions. The CPU then derives control signals for each of the reel motors using a multiplying, eight-bit D-to-A converter.

Complete alignment of the machine took about an hour for all bias/record/play adjustments. Some attempts to optimize the capstan and reel servo were tried upon encountering the high wow and flutter performance at 30 ips.

Reproduce and sync equalizations were aligned to the NAB curves for 7.5 and 15 ips speeds, and to the AES curve for 30 ips. Flux-level reference settings for all three speeds was 250 nW/m at 1 kHz using the tapes indicated in Table 1. Tape stock used for all of these tests was the 10.5-inch reel of 3M Scotch 226 supplied with the machine. The 226 was biased over peak sensitivity as follows for each of the three speeds: 3.8 dB at 10 kHz for 7.5, 2 dB for 15, and 1.2 dB at 30 ips. (These are the figures given in the MTR-12 service manual; no figures are given for other tapes.)

Record level settings and HF response adjustments were then completed, using the unit's VU meters as indicators. The LF COMP control was adjusted at a frequency of 100 Hz, at a discretionary speed of 30 ips. The manual directs adjustment of the LF COMP control until the reproducer output at 100 Hz matches the 100 Hz level indicated in Input Monitor mode. No speed for this adjustment is indicated, however. The result of this adjustment method is to show little difference between the compensated and uncompensated LF response at 30 ips.

Record phase compensation was also performed as directed by the manual for each speed. All alignment signals were supplied by external test generators.

Evaluation

The input impedances of each channel were measured between audio input connector pins 1 and 2, 1 and 3 and the normal mode, 2 and 3; these data are shown in Figure 1. The curves for channel 1 only are shown, as they proved similar to those for channel 2. (Only the magnitude of the impedances measured from pin #1 (shield) to pins 2 and 3 are shown to avoid clutter.) The magnitude and reactive phase of the normal mode impedance between pins 2 and 3 are shown, since they are the most important in considering the effects of

Figure 1. Otari MTR-12 input impedance versus frequency for three different drive conditions.

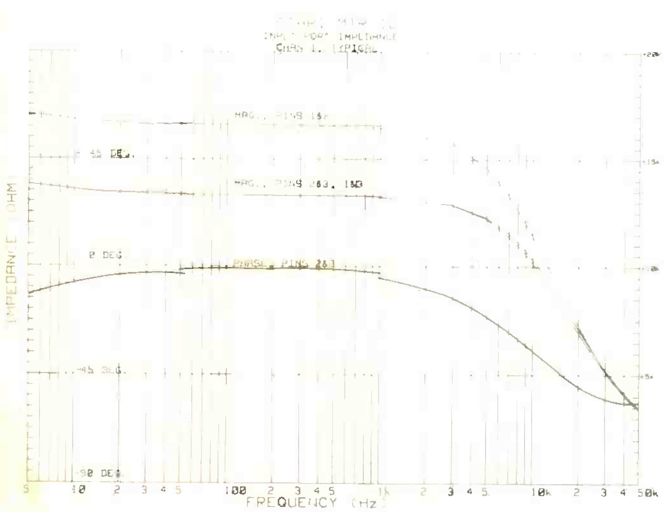


Figure 2. Output impedance versus frequency for three different drive conditions.

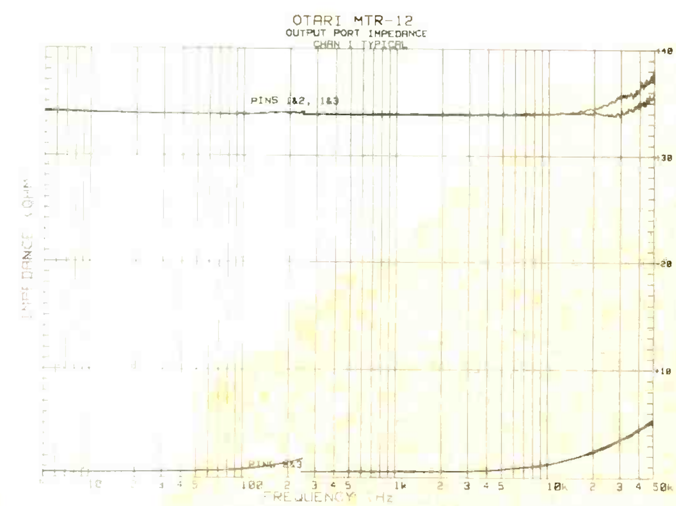


Figure 3. Logarithmic 20 Hz-43 kHz sweep showing input common mode rejection. The top trace is the -10 dBv input signal, the bottom trace is the output for the common mode feed. Vertical scale is 10 dB/div.

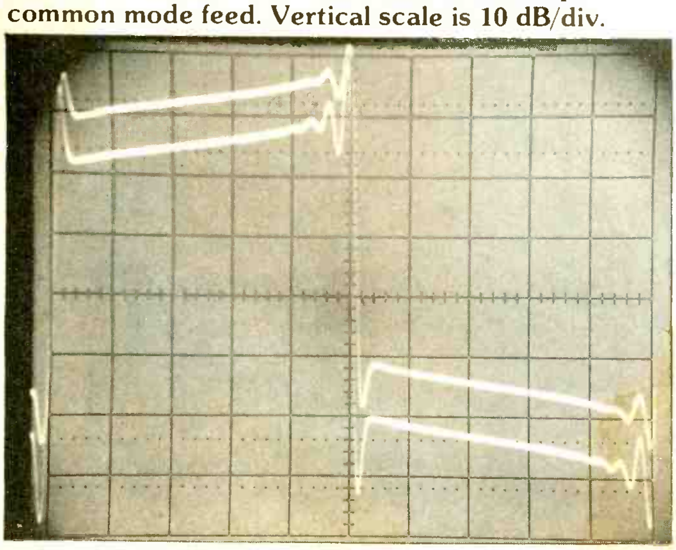
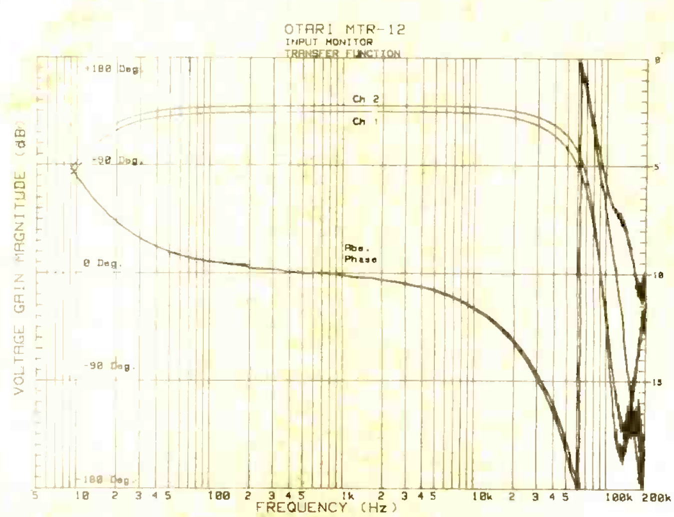


Figure 4. Signal chain transfer function for Line Input Monitor function. Load resistance is 10 kohms.



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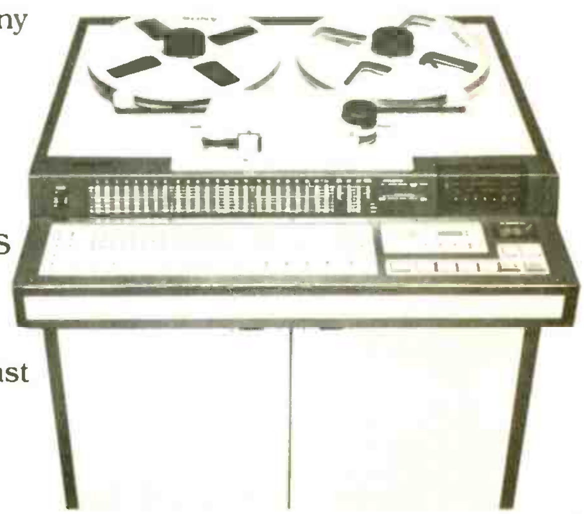

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interfacing the majority of devices. As can be seen from the data, the input port is differential imbalanced. The specified input impedance magnitude of 20 kohms could not be observed.

Impedance measurements of the two output ports showed very close similarity. As a result, the data for channel 1 output is shown as typical in Figure 2. Output impedance imbalance measured about 1.2%, implying a 38.6 dB common-mode rejection ratio for an ideal balanced load. A high-impedance, imbalanced differential input port looking out into this kind of load would be substantially balanced, as the output port impedances between the signal rails and ground (common) would be determined by the relatively low output impedance balance. The balance source impedance measured below 0.5 ohms for each channel. Clearly, any commonly-used load impedance could be driven from these ports with no difficulty.

The input common-mode rejection response is shown for both channels over a 20 Hz to 43 kHz range in Figure 3. The topmost, flat trace shows the sweep generator signal level of -10 dBv over the range. The degree of common-mode rejection remains greater than 50 dB below 43 kHz.

Calibration of the resident VU meters was checked by driving the input ports in Input Monitor mode, and adjusting the generator output amplitude to yield a constant +4 dBv at the individual channel output terminals. The meter readings taken under these conditions are shown in Table 2. The accuracy of the level indicators of audio equipment is a matter that most of us, this reviewer included, tend to take for granted. Past errors and regrets have prompted inclusion of indicator calibration data for the benefit of those similarly burned.

The Input Monitor transfer function is shown plotted in Figure 4. Data from the two channels are shown overlaid, and are quite similar in both magnitude and phase response. The consequential group delay data for these curves is shown in Figures 5 and 6, from which it can be seen that the two sets of group delay data are very similar over the passband. The low-frequency roll-off, commencing at about 40 Hz, has the consequence of causing considerable time distortion of low-frequency transients due to the very high signal propagation delay compared with those above 2 kHz.

The high value of delay — exceeding 5 milliseconds at 20 Hz — is quite audible on a comparative basis, and can be perceived as a softness in bass transient response. This steep rise in low-end group delay will, of course, be observed in the record/play transfer functions to be examined below.

A flux-loop sweep of the reproduce head, taken at the output of the head pre-amplifier, is shown in Figure 7, traces for channels 1 and 2 being overlaid. A similarly-derived flux-loop sweep of the record head is shown in Figure 8. The slight increase in slope

Figure 5. Group delay response versus frequency for Line Input Monitor function below 200 Hz.

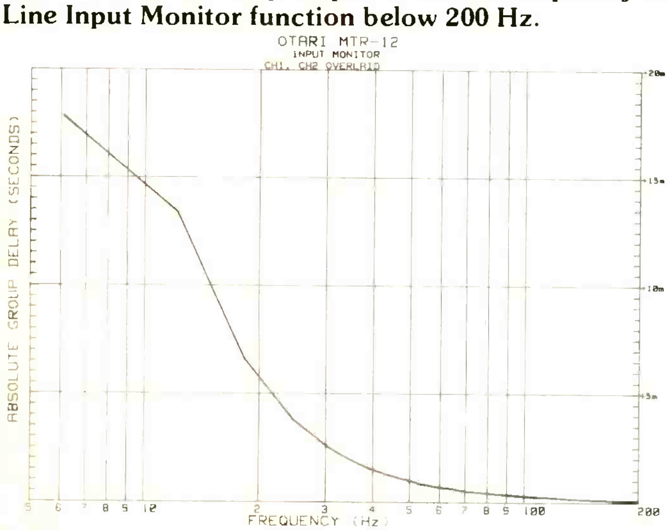


Figure 6. Group delay response versus frequency for Line Input Monitor function above 200 Hz.

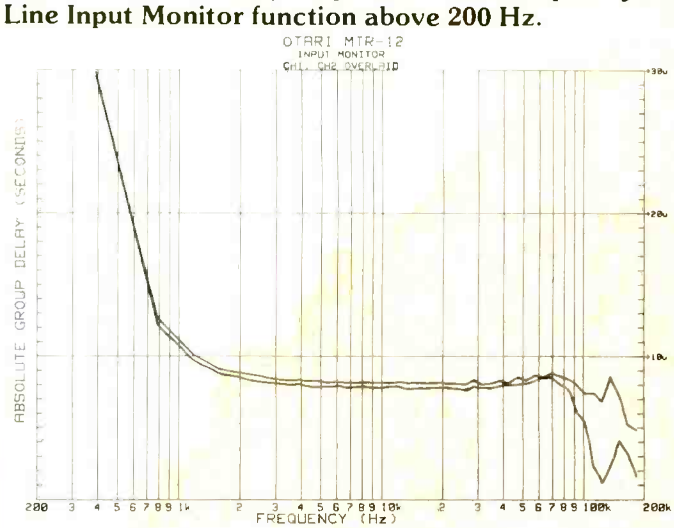


Figure 7. Flux loop response of reproducer head at output of the first gain stage. Channels 1 and 2 overlaid. Log frequency, 10 dB/div. vertical.

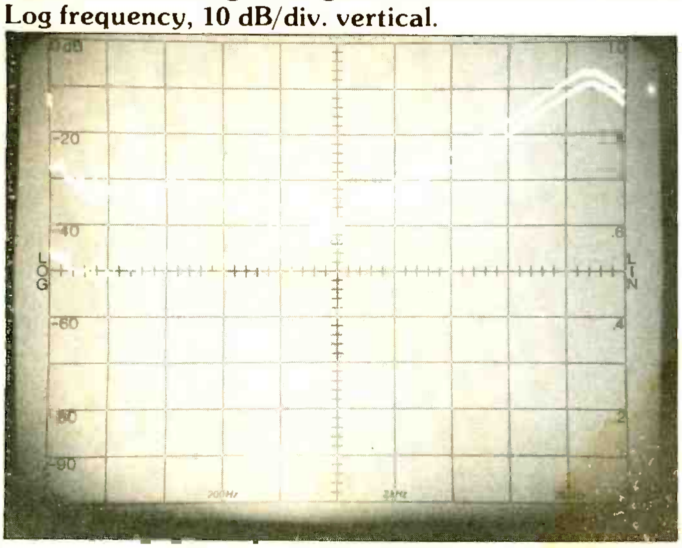


Figure 8. Flux loop response of the record head in Sync mode. Scale actors as for Figure 7.

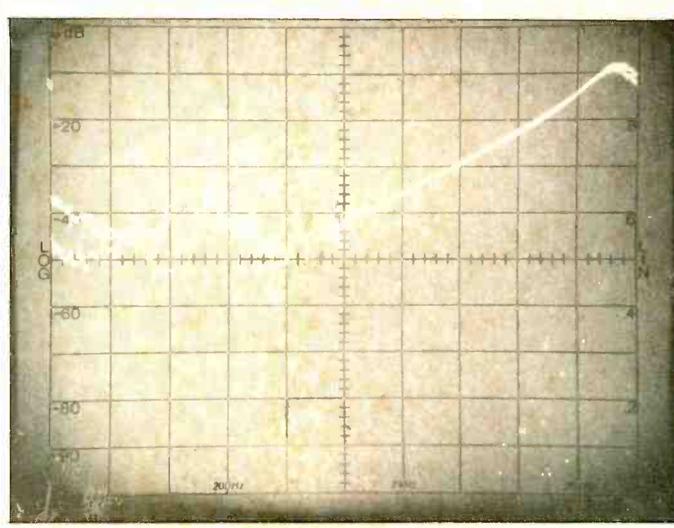


TABLE 1: SUMMARY OF OTARI MTR-12 SPECIFICATIONS

Description	Quoted	Observed
Track Config:	Two-Track, Half-inch	Yes
Tape Speeds:*	7.5, 15 and 30 ips. (190, 381 and 762 mm/s.)	Yes
Reel Capacity:	Up to 12-inch (305 mm); removable NAB hub.	Yes
Drive:	Servoed DC reel motors, and alternating pulsed-DC capstan motors; pinch-roller drive; 9.6 kHz speed reference.	Yes
Wow and Flutter :	30 ips: less than ±0.04%; 15 ips: less than ±0.05%; 7.5 ips less than ±0.06%; peak, DIN WTD.	±0.035% to 0.063%; ±0.027% to 0.051%; ±0.040% to 0.043%; peak, DIN WTD, W+F over four, 30-second sample intervals.
Pitch Control:	±20%; continuously variable	+23.7%, -24.7%; all speeds.
Speed Stability:	±0.03 max.	+0/-0.02% head reference to tail; 2,500 foot, 10.5-inch reel.
INPUT:		
Type:	Single op-amp input amplifier; differential, balanced.	Imbalanced differential single op-amp.
Impedance:	20 kohms balanced; +4 dBv nom. input	13.2 kohms across HI and LOW input port pins 2 and 3, 1 and 3. 16.4 kohms across input port pins 1 and 2; 1 kHz.
OUTPUT:		
Impedence:	Less than 5 ohms, active balanced, direct coupled; transformers optional; +28 dBv max output	Less than 0.5 ohms source impedance; ±1.2% of true balance.
Bias Frequency:	250 kHz	249.997 kHz.
Calibration Levels:	Switchable: 185/250/320 nW/m.	LOW / MID / HI Ch.1: -3.7 / 0.0 / +1.4 dB 170 / 260 / 305 nW/m Ch.2:-3.6 / 0.0 / +1.4 dB 172 / 260 / 305 nW/m (Data shown for 260 nW/m reference.)
Frequency Response: *Record/Play:*	**@ 15 ips** 18 Hz to 25 kHz; +1/-2 dB **@ 30 ips** 33 Hz to 27 kHz; +1/-2 dB	15 Hz to 26 kHz; +2/-3 dB.** 27 Hz to 31 kHz; +1/-3 dB.**
Frequency Response: *Record/Sync:**	**@ 15 ip** 30 Hz to 12 kHz; ±3 dB **@ 30 ips** 60 Hz to 20 kHz; ±3 dB	21 Hz to 14.4 kHz ; +0/-3 dB. 35 Hz to 28.5 kHz; +0/-3 dB.
Crosstalk:	Less than -60 dB @ 1 kHz.	Not measured
Depth of Erasure:	Greater than 80 dB @ 1 kHz.	Ch.1: -76 dB; Ch.2: -78 dB; 30-mil wavelength, from 520 nW/m flux level; 3M Scotch 226 tape.
*S/N Ratio, unWTD: (30 Hz to 18 khz; @ 3% third harmonic distortion to noise floor.)**	15 ips: 73 dB; 30 ips: 77 dB.	Ch.1: 62.1 dB, Ch.2: 63.0 dB; Ch.1: 62.9 dB, Ch.2: 64.1 dB; (20 kHz bandwidth, unWTD, 320 nW/m reference.)**
Internal Oscillator:	100 Hz and 1 kHz, sine; 1 kHz and 10 kHz squarewave.	Frequencies within 5% of nominal; levels trimmable.
Additional:		
Signal Interface:		3-Pin, XL-type connectors; pin #3 taken as "HI." Line In/Out: erect. Record/Play: erect. (Flux polarity in agreement with proposed standard if audio port pin #3 is taken "HI.")
Effectively Reproduce Head Gap Length:		Ch1: 298 micro-inch; ±10 Ch.2: 280 micro-inch; ±10
Effective Record Head Gap Length:		Ch.1: 556 micro-inch; ±25 Ch.2: 556 micro-inch, ±25
Flux-Frequency References:		MRL Test Tapes: 31T218, 31J219, and 31L220.
Shipping Weight:	200 to 220 pounds (91 to 100 kg).	
Power:	100 to 240 VAC, 50/60 Hz; 180W; IEC power cord.	

Manufacturer: Otari Corporation, 2 Davis Drive, Belmont, CA 94002. (415) 592-8311.

EDITORIAL NOTES:

*Otari does not specify frequency response and signal-to-noise ratio at the 7.5 ips speed for its half-inch two-track, since the MTR-12 is optimized for high-speed (15 and 30 ips) operation.

**Our reviewer apparently caught sight of an alternative specification sheet of frequency response tolerances for the MTR-12, hence the lack of correspondence between the quoted and observed response limits.

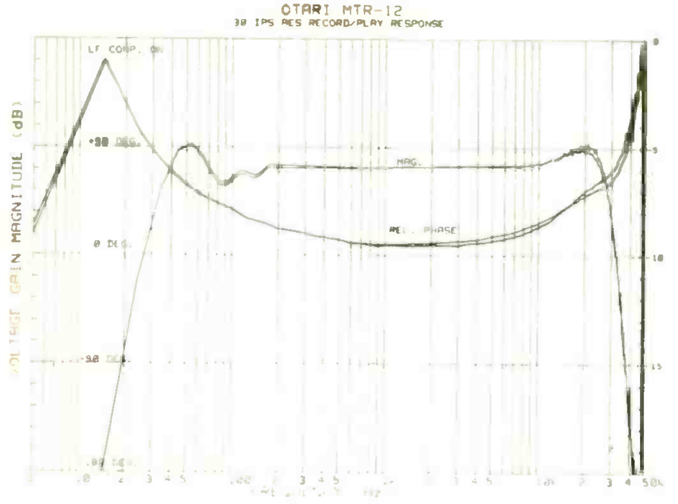


Figure 9. 30 ips record/play transfer function. Channels 1 and 2 overlaid.

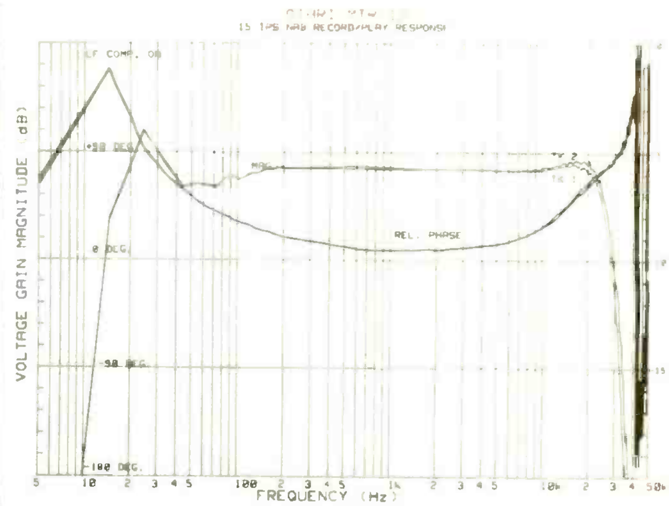


Figure 10. 15 ips record/play transfer function. Channels 1 and 2 overlaid.

as the peaks are approached from below indicates that the heads are slightly underdamped. The respective resonant frequencies are very nearly identical. This detail is important in the preservation of phase (not polarity) coherence between channels, and improvement in audio signal intelligibility, transient response, and stable stereo imaging of the reproduced signal information. In fact, for the case of a tape machine where phase compensation is provided, as with the MTR-12, failure to critically damp the reproduce heads makes adjustment of the phase compensation nearly impossibly. This is because most of the observed leading-edge squarewave overshoot used as the criterion for optimal phase compensation is due to the response of the undamped reproduce head resonance, and cannot be easily separated from phase-induced overshoot or other unwanted amplitude response.

It should be noted by the reader that this careful matching of the reproducing head to its electronics is a very critical matter, and will make optimal interchange of heads and head assemblies more than a matter of merely replacing a few pieces of hardware, followed by a casual re-alignment of levels, azimuth and bias. Electrical characteristics of the MTR-12's reproduce and sync head signal input ports are governed by fixed components. It would be a bit easier to manage head exchanges if the terminating resistor were adjustable over some range, and the resonance-determining capacitors made easier to change without the risk of damaging the circuit board. The importance of trimming the capacitor value for matching head resonating frequencies is not terribly important if the inductance tolerances of the heads and the lengths of the head interconnection cables can be closely controlled.

Optimization of the reproduce-head interface is a task that should not be undertaken without a flux loop, a low-frequency, swept, spectrum analyzer, and some numerical computation capability. As valuable as the fruit of this effort may be, it should only be undertaken by those competent and sufficiently equipped and motivated to do so.

While the service manual provides no mention of the need nor technique for performing this kind of optimization, reference 1 and 2 are offered for guidance to the interested reader.

All of the transfer function data shown here has been derived using Fourier transform techniques taken from the digitized time-domain system response. The record/reproduce electronics were aligned as described above, and the time-domain data taken for both channels. The reason for this note is that the magnitude response data shown in Figures 9, 10 and 11 result from a continuous-wave

Table 2: Otari MTR-12 VU Meter Calibration versus Frequency (Levels at output ports held constant.)

Frequency (Hz)	Channel 1 (dBv)	Channel2 (dBv)
10	-0.2	-0.15
20	-0.05	-0.05
50	0.0	0.0
100	0.0	0.0
1 k	0.0	0.0
2 k	0.0	0.0
4 k	0.0	0.0
8 k	+0.05	+0.05
10 k	0.0	0.0
12.5 k	0.0	0.0
16 k	0.0	-0.05
20 k	-0.05	-0.10
25 k	-0.2	-0.3
30 k	-0.3	-0.5

signal having odd harmonics regularly distributed throughout the frequency bank of interest. All odd harmonics of the modulating signal are present at all times. The MTR-12 was aligned using pure harmonic tones, since this is the easiest method to implement, and the most common approach by far. The data resulting from a continuous sweep of sinusoidal signal is a far different physical and practical matter from the response of an analog recording system to a continuous spectrum signal.

Figure 9 shows the transfer function at 30 ips, with channels 1 and 2 overlaid. The effects of the slight underdamping of the reproduce head can be seen as a slight peaking of the 30 and 15 ips magnitude response around 20 kHz, seen in Figures 9 and 10, respectively. Phase compensation has been adjusted as judged optimal, and the LF COMP feature remains active for the curves shown for all three speeds. The curves for the sync mode have not been included here, since they are rarely relevant for the case of a two-track machine. It is uncertain how representative the performance of the two-track, half-inch format is compared to the four-track situation in this respect.

The high-frequency magnitude response shown in the 7.5 ips curves of Figure 11 demonstrate a significant and noticeable error from the flat continuous-wave sine response noted upon initial alignment. The 7.5 ips frequency response alignment was performed at 10 dB below the 1 kHz reference flux level.

Figure 12 shows the relative group delay for the case of channel 1 record/play, for the three speeds overlaid. The group delay rises to a peak value of nearly 15 milliseconds at 20 Hz for the 7.5 and 15 ips cases. The 20 Hz delay is about 10 milliseconds for the 30 ips case where no LF pre-emphasis

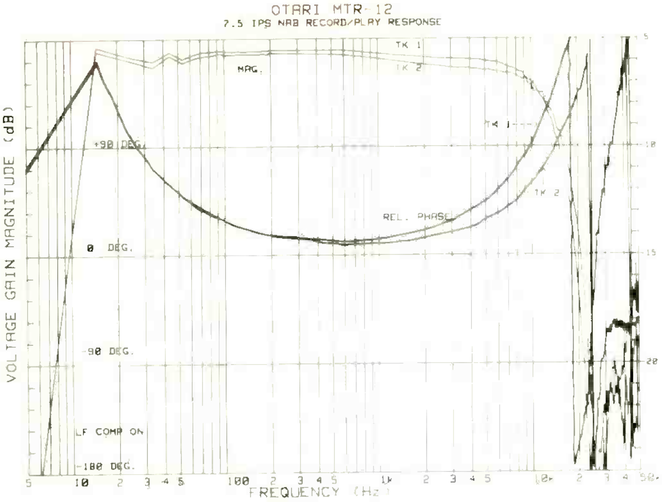


Figure 11. 7.5 ips transfer function. Channels 1 and 2 overlaid.

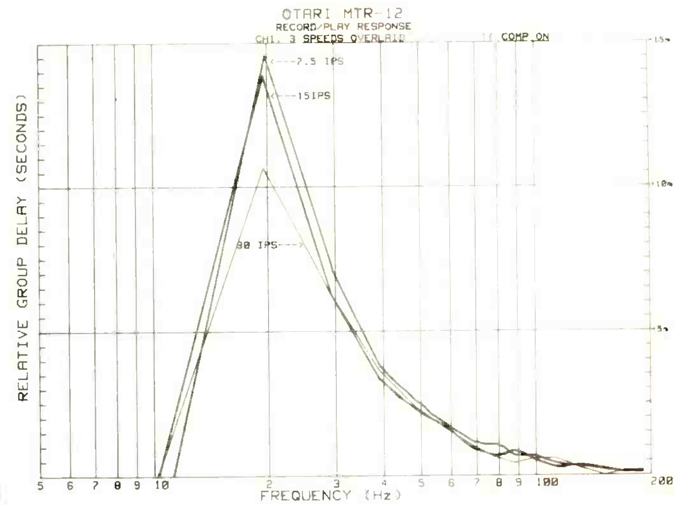


Figure 12. Channel 1 relative group delay characteristics for record/play transfer functions below 200 Hz. Three speeds overlaid.

is used. The implied 5 millisecond contribution from the audio channel electronics would seem to be a major source of this machine's low-frequency delay time characteristics.

The relative group delay response of channel 1 above 200 Hz for the three speeds is shown in Figure 13. The curves have been deliberately separated for clarity, so the vertical scale values should not be taken too literally. The matter of importance to be noted here is the fact that the delay responses are essentially flat above the 1 to 2 kHz region. The onset of low-frequency delay could be shifted farther toward the left side of the scale by careful selection of interstage coupling components. The sharp dip in the 7.5 ips delay curve of Figure 13 is caused by the reversal in polarity (not phase) of the reproduced signal in the vicinity of the gap null at about 25 kHz.

The effects of the MTR-12's LF COMP and PHASE COMP features are shown in Figure 14 for the case of 30 ips record/play, channel 1. Given the adjustment procedure of the Otari manual, there seems to be only minor difference between use of the LF COMP feature and not. Extremes of LF COMP adjustment were not examined. The right-hand side of Figure 14 shows the optimal and extremes of the record phase compensation control for the 30 ips case.

Figure 15 shows the effects of the extremes of phase compensation adjustment on a 1 kHz squarewave at 30 ips for channel 1; the pre-transition overshoot can be adjusted to be outrageously high if one desires. The effects of phase response adjustment on the magnitude response is negligable. Therefore, phase compensation can be saved for the final alignment step without need to touch-up previous adjustments.

The MTR-12 noise performance comes in close to its specification. The difference is less than about a decibel, and could be attributed to variations in visual judgement on the part of the observer.

The flutter data cannot be attributed to any such vagary of perception, however. Careful verification of this writer's flutter meter calibration using a frequency-modulated signal, a spectrum analyzer, and a table of Bessel functions show that the observed flutter data are accurate. The data in Table 1 quoted were derived by recording the flutter meter 3.15 kHz reference signal for a five-minute period, and then rewinding the tape to the beginning of the reference recording.

During the recording, the flutter indications were noted for reference, and were typically half the specified peak DIN-weighted numbers, or less. Upon replay of the tones, the flutter

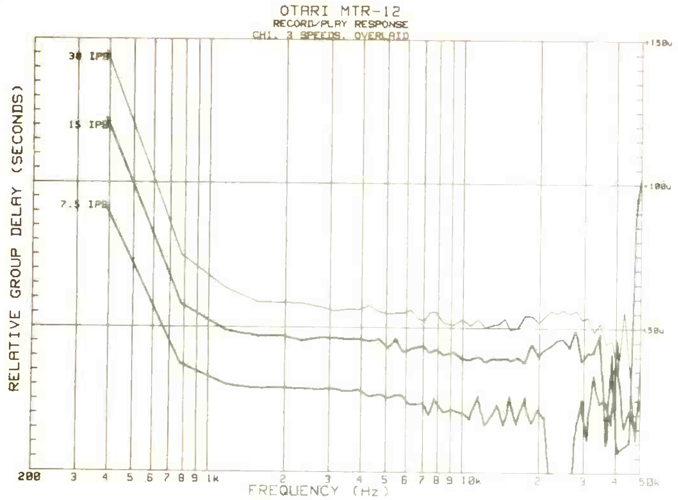


Figure 13. Channel 1 relative group delay characteristics for record/play transfer functions above 200 Hz. Three speeds overlaid.

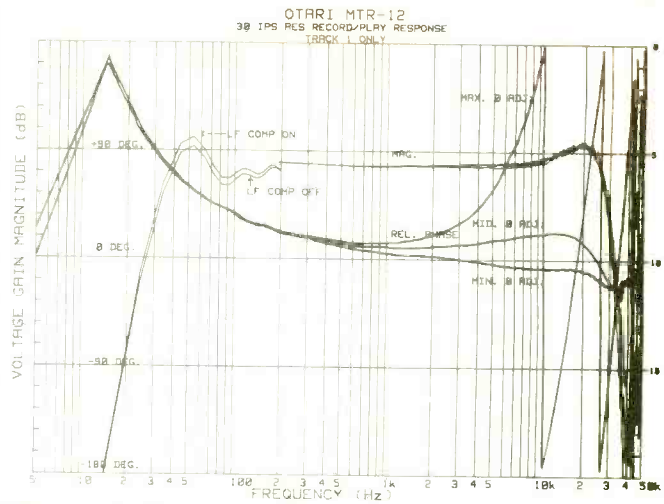


Figure 14. Channel 1 30 ips record/play transfer function. Extremes of record PHASE COMP adjustment and LF COMP are shown overlaid.

invariably read much higher than during the recording. The culprit, especially in the case of the 30 ips data, seems to be a 27-Hz flutter component that can rise as much as 20 dB over the residual flutter signal spectrum central skirt on replay. This component shifts to about 13 Hz for the 15 ips measurement, and to around 7 Hz for the 7.5 ips case. The action of the DIN-weighting curve reduces the significance of this component as its frequency declines; lower speed flutter performance therefore improves on a weighted scale.

Appearances are that this component is hidden during playback monitoring of the flutter signal while recording, because the distance between the record and play head gaps seems to very closely approximate to the circumference of the capstan shaft. The troublesome 27-Hz component would be expected to be the same as the rotational frequency of the capstan shaft. The data given in Table 1 are the results of four, 30-second samples of replayed flutter displayed as a peak-hold indication for each of those 30-second intervals.

The entire MTR-12 system is polarity-consistent. The reproducer polarity (not phase) response conforms to the standard proposed by Lipshitz and Vanderkooy[3] in 1981, provided that the audio connector pin #3 is taken as the "HI" port terminal.

Summary

The MTR-12 seems to be sturdily constructed. The deck is constructed from a sheet of quarter-inch aluminum, and rests within a welded steel frame when console mounted. The head assembly is rigidly constructed, and azimuth adjustments are made without disturbance of the head height. All controls are plainly marked. The mechanism is quiet in operation, and tape handling appears to be gentle.

Circuit-board assemblies are ruggedly constructed, with components identified by silk-screened designations on the conductor side. Bold-plated board and edge connectors appear to be used throughout. Use of ceramic capacitors within the audio signal chain appears to have been avoided. There seem to be no use of unbiased Tantalum capacitors within the audio chain.

Most adjustments are single-turn miniature trimmers that may prove to be fragile if adjusted forcefully, or with inappropriate tools. A drop of epoxy to secure the trimmer body to the circuit board would be a reasonable preventive measure that would save frequent replacement of these controls.

Interfacing of this machine should present no problems, provided that the installation is designed with recognition of the significance of the

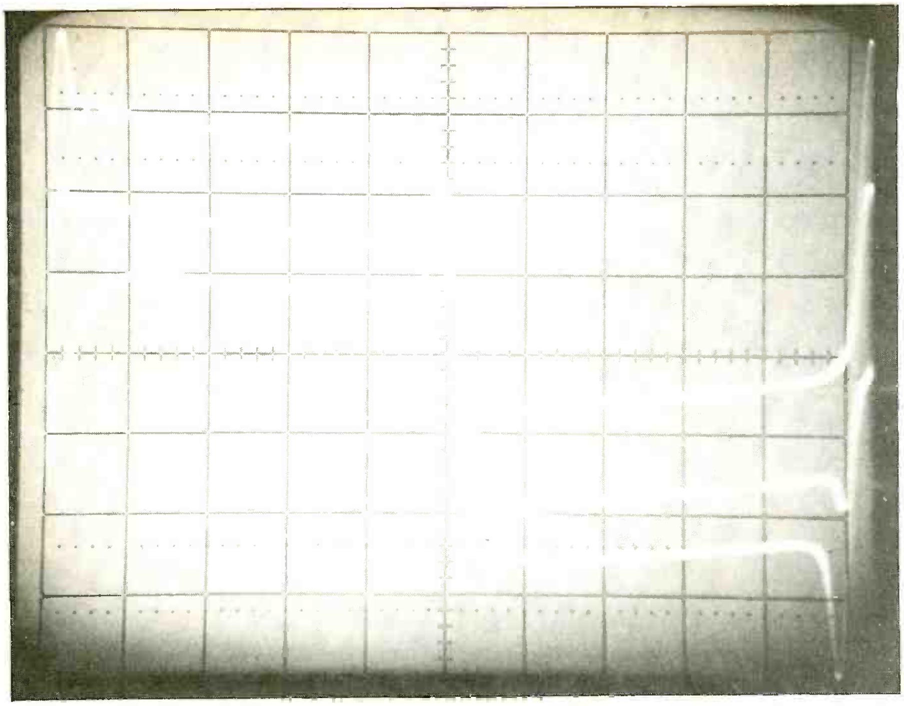


Figure 15. One kHz square wave response for extremes and optimal adjustment of the PHASE COMP control at 30 ips. These waveforms correspond to the phase data of Figure 14.

audio connector signal assignment. If the machine is arbitrarily connected to a system having audio connector pin #2 grounded, the "LO" side of the output ports will be shorted, with possible crosstalk and distortion problems the result. Activation of the "BAL/UNBAL" switch that disconnects the LO output driver, with a consequent 6 dB loss in maximum output level, is necessary for optimal performance in an unbalanced situation.

The drop in input impedance above 10 kHz could have some impact if the source impedance of a device feeding the MTR-12 is not of sufficiently low value. Conditional instability on the part of a high source impedance driving system could also have disappointing results.

The only real problem that this machine exhibits seems to be the stubborn nature of the 30 ips flutter performance discussed above, and would appear to be a fundamental problem that should be addressed by the manufacturer. Overall, the specified flutter performance of the MTR-12 is about twice as high as I have been accustomed to dealing with in day-to-day audio affairs of life. ■■■

References

1. "Flatter is Fatter — Magnetic Reproducer Equalization Accuracy," P. Butt, *Recording Engineer/Producer*, October, 1978, Vol. 9, No. 5, p. 84.
2. "ATR-100 Series Recorder/Reproducer," service manual. Ampex Corporation, Redwood City, California, catalog no. 4890407-01, 1977, pp. 5-13 -5-21, 5-25.
3. "Polarity and Phase Standards for Analog Tape Recorders;" J. Vanderkooy and S. P. Lipshitz; AES Preprint 1795 (D-4); May 12, 1981, Los Angeles.

Our thanks to Everything Audio, the Encino, California-based Otari dealer, for its valuable assistance in obtaining the MTR-12-H for this review — *Editor*.

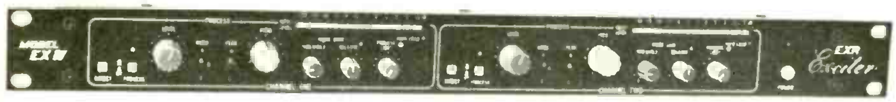
MANUFACTURER'S COMMENTS

Otari's national sales manager, John Carey, responds to the above technical assessment as follows:

"We are very impressed with the level of technical expertise evident in this report, and are pleased that *R-e/p* has adopted this new format of technical review. The MTR-12 Series of tape machines historically has benefitted from this type of close analysis, and the MTR-12 Series II, which will be introduced at the AES Convention in October, incorporates feedback from many of our users.

Two points regarding the machine reviewed should be made for clarification:

1. With regard to the observed monitor circuit bandwidth, the alignment input jack on the front panel was designed for use with test tones, and is bandwidth limited for that purpose.
2. With regard to the 7.5 ips performance, the MTR-12-H was optimized for high-speed (15/30 ips) half-inch/two-track mastering operation. The benefits of our decision to compromise in favor of superior high-speed performance on this model are obvious. Otari warrants all of its products to meet or exceed all published specifications when delivered." □□□



EXR EXCITER MODEL EXIV PSYCHO-ACOUSTIC AUDIO PROCESSOR

Reviewed by Roman Olearczuk

The latest EXR Exciter Model EXIV is a fourth-generation device evolving from continual research and an on-going communication with recording engineers and mixers in the field. This aesthetically pleasing package provides the user with greater control over the enhancement process by providing a variety of new features, including a variable notch frequency control, an adjustable process noise-gate and limiter, and a selectable enhancement peaking switch. A new process setting — Mode A — also has been added to deal with the bass frequencies. Balanced XLR-type connectors and bar graph indicators are now included for easy interface and setup.

Overall, this new model is quite an improvement over the EXIII. But the same questions still remain: What is the unit supposed to do, and does it really work?

The mystery in the design of EXR's psycho-acoustic process is inherent in the company's steadfast policy of not disclosing circuit diagrams or complete theory details. The actual processing circuit is still contained within an epoxy-encapsulated module; future editions, the company says, may even have their own custom ICs for better trademark secrecy. All that could be discovered about the internal workings of the unit is that somewhat abstract technical description to be found in the product literature: "Each of the three enhancement process settings (A, B, and C) in the EXIV are a precise variation of the integral functions performed within the EXR Exciter:

- Pre-selective 180-degree phase notching,
- Time manipulation,
- Frequency manipulation, and
- Psychoacoustic juxtapositioning.

"All processing of program material by the EXR Exciter is performed in a side chain. The first three functions create an interference signal which, when added back into the original signal, reverses the primary or fundamental build-ups and losses caused by multiplier effects and distortion in the audio reproduction chain. 180-degree phase notching is used to cancel out specific frequencies where distortion tends to build up. By notching these unnatural build-ups, the ear's internal limiters are relaxed, causing an apparent (to the ear), but non-measurable increase in the sound pressure level of the overall frequency spectrum. Even though there is an apparent volume increase, when used correctly the EXR Exciter does not affect the actual volume level because it actually cancels out as much as it adds.

"Psycho-acoustic juxtapositioning is an exclusive EXR process that appears to fill the holes left by the 180-degree phase notching. Information is extracted from one part of the spectrum, processed, then used to sonically replace another part of the spectrum which has been eliminated or lost."

At the time of writing, the above information is all that EXR wants to reveal about the internal workings of the Exciter. Otherwise, the device was tested during daily television promotion routines at NBC Burbank, and this hands-on report concentrates on those operational and qualitative results.

SUMMARY OF MODEL EXIV SPECIFICATIONS

Maximum Input/Output Level:
+27 dBV (Level switch 0 dB);
+7 dBV (Level switch -20 dB).
Input:
Instrumentation amp type balanced input;
Impedance: 40 kohm (balanced)
20 kohm (unbalanced).
Output:
Transformerless AC floating circuit;
Impedance: 150 ohm.
LED Level Indicator: (indicates input level on process circuit only) -20/+6 dBV, 12-segment.
Noise Gate: Threshold range
-40 to -5 dBV (level switch 0 dB);
-60 to -25 dBV (level switch -20 dB).
Process Limit Level: Range
-10 dBV to infinity (level switch 0 dB);
-30 dBV to infinity (level switch -20 dB).
Power Supply:
U.S. 117V AC 50/60 Hz, 10 VA; Internally switchable to 230V AC, 50/60 Hz.
Signal to Noise Ratio: Better than 90 dB.
Dimensions: 19-inch rackmount, 1¾ inches high by 10 inches deep. (48.3 × 4.3 × 25.4 cm).
Price: $1,690.
Manufacturer: EXR Corporation, 3373 Oak Knoll, Brighton, MI 48116. (313) 227-6122.

Input/Output Characteristics

The EXR Exciter Model EXIV has two independent, identical channels, each processor being easily accessed through the use of standard XLR-type connectors. The input is a balanced electronic circuit with an impedance of 40 kohm, while the output is a transformerless, AC floating circuit with a 150 ohm impedance. A Set Level switch located on the rear panel allows the user to add 20 dB of attenuation to both channels for direct instrument or microphone applications. As stated in the Owner's Manual, this Set Level switch does not affect the unity gain design of the Exciter. In the -20 dB position, only the signal entering the Processing Module circuitry (see accompanying block diagram) is amplified 20 dB for proper level conditioning. After the signal leaves the Processing Module, it is then attenuated 20 dB for correct level mixing with the original signal.

The input level to the unit must be set externally. However, the EXR comes equipped with two, 12-segment LEDs that serve as input bar-graph level meters, and provide a visual indication of signal strength over the range of -20 to +6 dBV. The user sets the signal being fed to the EXR for an optimum reading of 0 dBV. Peak signal levels can exceed this value but should not trigger the red Overload LED set for +21 dBV (+1 dBV with the Set Level switch at -20 dB). Maximum EXR input/output level is +27 dBV (+7

OPERATIONAL ASSESSMENT
EXR EXCITER MODEL EXIV

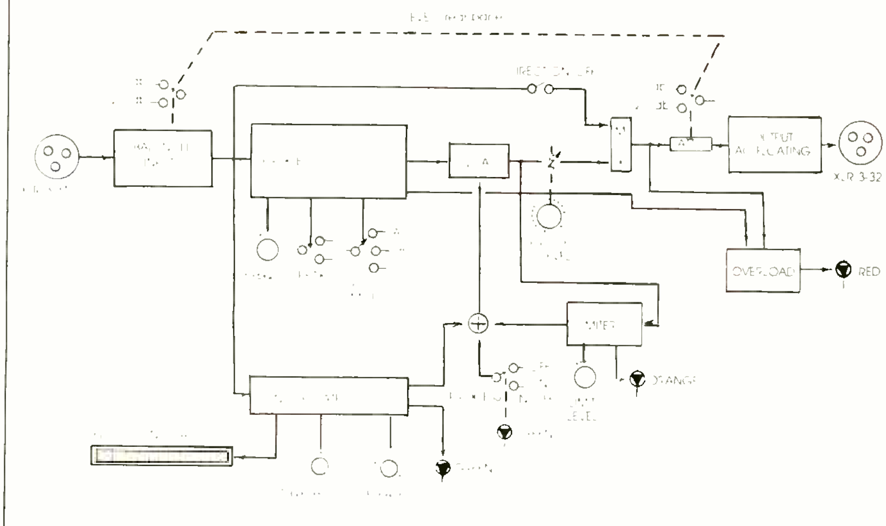
dBV with Set Level at -20 dB).

Controls Description

Each channel has the following process controls (see accompanying front-panel layout): Direct on/off select; Process on/off select; Level control; A-B-C Process select; variable Freq control, Peak H/L frequency Select; Noise Gate Threshold and Release controls; and Process Limit adjust. Apart from the Direct signal on/off select, all other front-panel controls manipulate the signal routed through the processor circuitry. It is this processed signal that gives the listener with the effect of increased presence, clarity, and separation within the mix, but these improvements in sound only occur when the original signal is combined with the altered signal.

The unit provides the user with a choice of operational modes: Process and/or Direct. With both pushbuttons off, no signal appears at the output. When the user engages the Process button, only the altered signal via the process Level control appears at the output; a green LED indicator alerts the user that this process signal is being routed to the output. If the Direct button is also engaged, the original signal at unity gain is mixed along with the desired amount of processed signal. In the Direct mode, only the original signal appears at the output.

Block Diagram
EXR EXCITER Model EXIV



The A-B-C Mode switch provides the user a choice of three EXR enhancement processes within three overlapping frequency bands. Although each band extends to 20 kHz, it is the low-end of the frequency spectrum that is treated uniquely within each mode. The A-Mode accentuates the bass frequencies starting around 60 Hz, while the B-Mode starts emphasis in the mid-range about an octave higher. The C-Mode provides enhancement in the high frequencies yet another octave about the B-Mode spectrum. A continuously variable frequency control, labelled Freq, provides an adjustment of the center frequency within each selected mode. A Peak H/L switch is a bandwidth (Q) control for accentuating the center frequencies: Peak H is a high-Q/narrow-band emphasis, while Peak L provides low-Q/broadband boost, and is considered the normal setting.

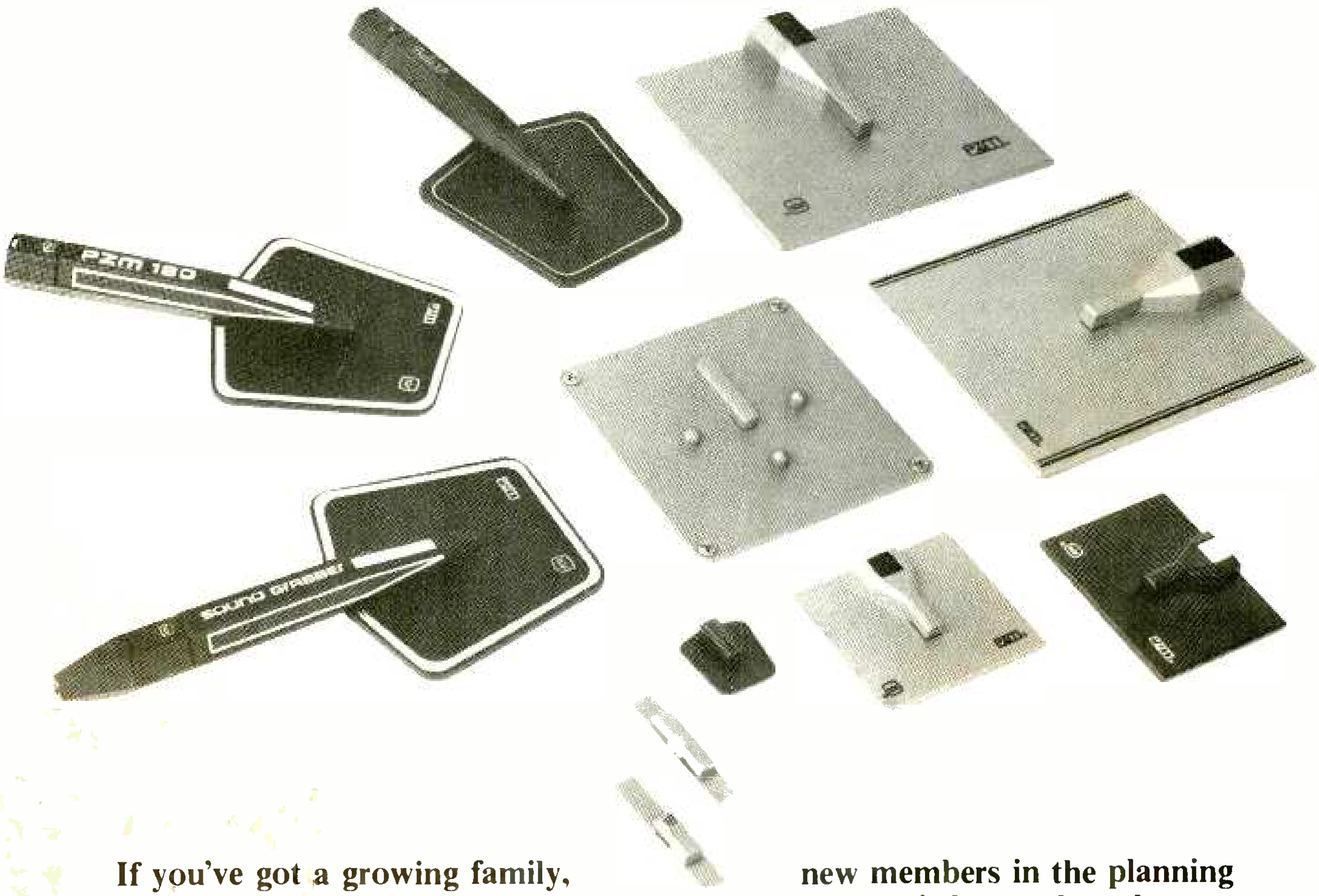
The Noise Gate Threshold and Release controls help to reduce noise and hiss in the processed signal only. The Threshold range extends from -40 to +10 dBV (Set level switch at 0 dB); a green LED provides an indication when the signal exceeds the Threshold setting. The logarithmic Release control varies release times from 120 milliseconds (labelled "0") to 5 seconds (labelled "10"). The attack time is fixed internally at an optimum value of 0.32 milliseconds per dB.

The Process Limit adjust limits the process signal from high-frequency spikes, and control provides a constant compression ratio of approximately 14:1 over a variable threshold from -10 dBV to maximum signal input. An orange LED gives a visual cue that limiting is occurring above the Threshold level set.

Operational Comments

The Owner's Manual and Applications Guide provides set-up examples through illustrated front-panel drawings. Applications described include: basic tracking, multitrack mixdown,

Family Portrait



If you've got a growing family, sooner or later you need a picture with everybody in it. It's a statement of family pride, and we humbly admit that we are pretty proud of this group.

There was a time when most people didn't recognize a Crown PZM® as a microphone - even when they looked at one. Times have changed. Billboard Magazine reports in their most recent brand usage survey that 37.5% of U.S. recording studios use Crown PZMs.

This sort of demand, multiplied by many other applications, has made the family grow, with new microphones tailored for new users. In fact, the number of new members in the planning process is larger than the number in the picture. Since a lot of our friends have only used one or two models so far, we thought we'd better introduce the family. The next time we may not be able to get them all in one picture.

Keep an eye on this family. Right now it's one of the newest and best. It just might get to be the biggest.

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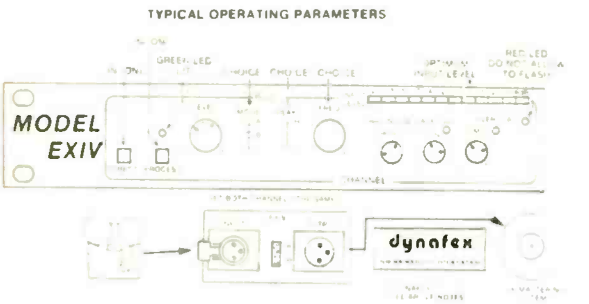
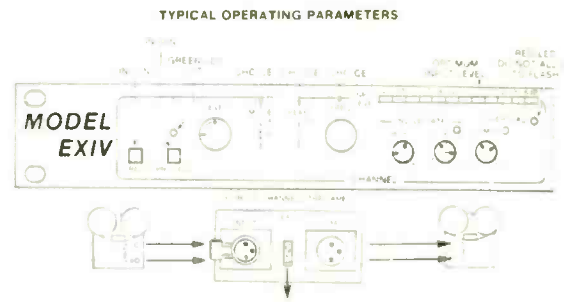
For additional information circle #115

APPLICATION OF EXIV IN DISK MASTERING



APPLICATION OF EXIV IN TAPE-TO-TAPE TRANSFERS



(Note: Dynafex™ is a trademark of MICMIX Audio Products, Inc.)

OPERATIONAL ASSESSMENT EXR EXCITER MODEL EXIV

disk mastering, tape transfers, musical-instrument enhancement, and AM/FM broadcast signal chains. Two applications — multitrack mixdowns and tape transfers — provide the basis for the operational comments within this review.

For mixdown, the Exciter EXIV can be used in either the side-chain effects return mode, or the in-line overall mix mode. The side-chain effects return mode provides the user with the most flexibility and control. For this function the EXR Process switch is in (on) and the Direct button out (off). As discussed previously, with these settings, only the processed signal is present at the output. This output signal is introduced back into the final mix either through an effects return, or an open console module.

The input signal feed to the unit is patched from either a console effects or cue send. Individual tracks (dialog, music, etc.) are sent to the EXR via the respective console sends so that the Input Level indicators are at an average reading of "0." The Level control is adjusted so that the output signal returns to the console at -4 VU. (Note: the EXIV's Input Level indicators are calibrated in dBVs and *not* VU; hence the 4 dB difference in level reading at the console return.)

Additionally, the process Limit control is set at maximum, so that the limiter is inactive. At this point the process signal is set for unity gain through the unit. Since the Model EXIV is a two-channel unit, signals to be enhanced can be treated independently and returned to center, or split left and right for stereo.

Once the internal gain structure has been set, fine adjustments in frequency, gating, and limiting can be made. For network television promotional spots, the EXIV was used primarily on announcer tracks. The process return was first brought into a mix at a -15 level on a console fader. Initial frequency control settings on the EXR were: A-Mode, L-Peak, and center Freq. The noise gate controls, Threshold and Release, were first set at "0" for minimum signal trigger, and quick release.

To increase clarity and presence, more EXR process was slowly added via the console fader until there was an awareness of a high-frequency "spitting" sound. When this point was reached in the mix, the process was backed off a slight amount, thereby yielding an appropriate enhancement setting. Different announcers required adjustments in Mode and Freq controls. Also soft and muddy readings were improved by switching to an H Peak mode, as well as lowering the Limit threshold. This control kept the sibilant "spitting" sound down as the upper mid-range presence became more emphasized.

Tape and disk transfers are quite a common occurrence during daily production routines. The EXR EXIV was used in-line, patched between the tape machine or turntable pre-amp outputs and the console line-input returns. Both the Direct and Process buttons were engaged. Initial settings were: Level at "3," Mode-B, Peak-L, Freq at "3," Threshold at "3," Release at "5," and Limit at "8." Input level averaged 0 dBV, and the Overload LED was not triggered on peak program material.

As different tapes were previewed prior to transfer, adjustments were made to the initial settings. For example, tapes that featured vocals required emphasis settings similar to those used on the announcer's voice. Muddy tapes required more process via the Level control. The Peak switch at "H" also added clarity.

Noisy disks were first treated with a KLH Transient Noise Eliminator to reduce surface pops and ticks. The process Noise Gate Threshold was adjusted carefully to eliminate as much of the remaining crackles that were being emphasized by the EXR process. Noisy tapes were treated the same way, except the program was routed through a MicMix Dynafex single-ended noise reduction unit prior to the EXR enhancement process. Signals were monitored with and without the EXR, and adjusted to taste. In all cases, the EXR EXIV helped rather than hindered the overall sound of these tape and disk transfers.

Although other applications were not tested for this assessment, one can see that the EXR would be useful for enhancing electronic instruments, and has applications during disk mastering. Guitars and keyboards can be patched directly into the unit after a few requirements are satisfied:

- The instrument cord must have a male XLR-type connector (pins 1 and 2 Lo; pin #3 Hi) for the EXR input;
- The Set Level switch must be set at -20 dB; and
- If the output of the EXR is feeding an unbalanced input (i.e., pre-amp section of instrument amplifier), a low/high matching impedance transformer or instrumentation amplifier would be required for proper interface.

As for disk-mastering applications, the Owner's Manual provides additional notes on the use of a Dynafex noise reduction unit while mastering noisy material. The recommended practice is to insert the EXIV before the Dynafex. Also the input connection to the EXR should be as direct as possible.

In general, the device worked exactly as advertised. The only criticisms I would offer for the EXIV are of a minor ergonomic nature. While all controls have labels (i.e., "0" to "10"), an indication of their actual operating values — such as -10 dBV (instead of "0") on the Limit control — would be more useful for the user. Also, as noted earlier, the "0" reading on the Input level indicators should match "0" on the console.

Conclusions

In this reviewer's opinion, the EXR Exciter Model EXIV is a unique and very useful audio device. In fact, only by momentarily bypassing the unit during a mix did I realize the full extent of audio enhancement provided by this latest generation unit. At times it was virtually impossible to duplicate the same clarity in a mix through equalization, without adding an unnatural sound. Unlike record mixing, television promotional spots have pre-mixed elements that have to be brought together to produce a cohesive sound. The music is a finished mix, and the added sound effects, at times, seem as though they will never fit without drastic equalization. (You cannot just EQ the bass guitar a little differently, for example, in order to hear the explosions better.) Add to this an announcer's voice and dialog tracks from various sources, and suddenly the whole mix is sonically fighting you!

The EXR EXIV helps immensely in creating "aural pockets" for these diverse sounds; the whole mix becomes clear, tight, and smooth. Presented with the choice of whether to use or not to use the EXR in a mix, I found myself always preferring the sound via the Model EXIV. ■■■

New Products

AMEK ANNOUNCES DIGITAL EDITING SYSTEM FOR SONY PCM-F1

Available in the U.S. from KEMA Marketing, a division of AMEK Consoles Inc., CLUE (Computer Logging Unit and Editor) is said to provide a cost-effective solution to the problems of using the Sony PCM-F1 and PCM-701ES digital audio processors in a professional recording environment, as well as providing an invaluable aid to operation for the PCM 1610 system.

CLUE exists primarily to satisfy the need for editing facilities on mastering systems employing the low-cost F1 and 701ES digital audio processors. In addition, the system provides logging and autolocation facilities, as well as more accurate counters than those found on domestic VCRs. CLUE enables butt copy-editing to be performed either in analog or digital modes to frame accuracy (1/30th of a second with NTSC video). The unit also facilitates the insertion of auxiliary devices into the signal path during edits, and makes provisions for fader and level adjustments.

The complete system comprises a 4U, 19-inch rack unit that houses the controlling computer, disk drive, switching circuitry and interface connections, and a remote typewriter keyboard. Communications with the user is effected through the VDU monitor, which displays available commands recorder status information, counters, and logging details. The program is entirely menu-driven with smart commands being one-key entries.

According to AMEK, plans are well advanced for further interfaces for CLUE, which will allow communications with a wide variety of audio and video recorders, as well as the ability to read and write timecode.

Retail price of the CLUE system is expected to be less than $8,000.

AMEK CONSOLES INC.

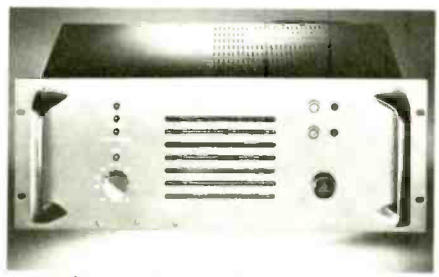
For additional information circle #119

FM ACOUSTICS FM1000 AND FM801 POWER AMPS NOW AVAILABLE IN U.S.

The FM1000 is described as the only non-bridged power amplifier that delivers continuous power in excess of 1000W, even into the most difficult loads. The new FM801 is a two-channel amplifier employing all the advantages of the FM1000.

Unique features are said to include a non-polarized, discrete input stage with a common-mode rejection better than 75 dB; completely discrete Class A circuitry throughout; and an overdesigned output stage that allows peak currents in excess of 100A, and continuous RMS currents in excess of 60A.

The FM1000 can drive any load between one ohm and 10 kohm, while the FM801 can drive down to 1.5 ohms per channel without any form of limiting or compression. Both amplifiers include purpose-designed 200A heavy-duty load connectors. A built-in protection circuit continuously checks all important parameters, such as ±DC offset, temperature on the cooling fins, temperature of the transistors, fan speed, HF signal presence, as well as output load impedance.



According to the manufacturer, this means that the amplifier will perform without *any* form of limiting into a one ohm or a 1.5 ohm load, respectively, and still immediately shut off should a short circuit occur.

Also available is the FM236 electronic crossover featuring fully discrete Class A circuitry, and proprietary linear-phase filters achieving 36 dB per octave attenuation.

FM ACOUSTICS USA

For additional information circle #120

ROLAND SBX-80 SYNC BOX FOR MIXED TIMECODE SYNCHRONIZATION

Intended to solve problems of varying synchronization codes used by film, video and electronic musical instruments, the SBX-80 is a multi—timebase, SMPTE- and MIDI-compatible clocking device. As a master controller, the unit can read a variety of signals, including audio click-track and live performance cues, while simultaneously sending synchronizing information to many other devices that use different timecodes.

SMPTE functions as the common denominator for all of the other timing codes. Using the system, each medium (audio, film, etc.) can be inscribed with a code that not only allows other mediums to perform in synchronization, but the code also enables the user to search, retrieve, insert, and delete individual sections.

The SBX-80 can accept input from SMPTE, DI, audio click, and its manual tap buttons; the latter allows the user to establish or change tempo in real time. A numeric value can be assigned to quarter- or eighth-notes using the unit's keypad. The Sync Box then sends through its outputs MIDI, Sync-24 (Roland's sync code), a programmable Time Base (1, 2, 3, 4, 24, 48, 64, 96, or 120 pulses-per-quarter-note), SMPTE, and a Metronome signal.

Recommended retail price of the new SBX-80 Sync Box is $1,195.

ROLANDCORP U.S.

For additional information circle #121

NEW PATCHBAY CLEANING TOOL FROM VERTIGO

Made to withstand the most severe abuse, the new Patch Bay Burnishing Probes are machined from hard steel to patch-plug specifications. The probe tips receive their burnishing properties from a special process, and the entire tood is then hard-chrome plated for durability.

The burnishing probes can be used to clean all forms of contamination, such as smoke, moisture, and corrosion. Use of the probes is described as being very simple: just insert and twist them. Model PB-TT is designed for TT patch bays, and PB-TRS for ¼-inch TRS patch bays.

VERTIGO RECORDING SERVICES

For additional information circle #122

TASCAM INTRODUCES MINISTUDIO PORTA ONE

Comprising a portable, battery-operated four-channel mixer/recorder, the unit's built-in four-track cassette deck runs at 1⅞ ips speed, allowing playback of standard stereo cassettes. In addition, the Porta One can assign one or all four input channels to any track, and features switchable dbx noise reduction.

An easy-to-operate clearly-marked panel, flexible functions, and lightweight packaging, is said to make the Ministudio convenient and easy to use. Typical users range from composers and lyricists, to field recording of interviews, news gathering for broadcast or cable TV, making videos, or shooting a film on location.

The Ministudio's input channel mute switch permits bringing channels into the mix without changing a setting. Each channel has a two-band EQ with center detent plus pan for use during recording, overdubbing and mixdown. A discrete four-track tape cue mix and separately adjustable headphone output level control are also featured.

The cassette transport section is FG servo-controlled for smooth, gentle tape motion with variable pitch control, and a counter with Zero Return for cueing. Positive settings and protection against damage or accidental control movements are said to be ensured with low-profile, easy-touch controls.

TASCAM, TEAC PROFESSIONAL DIVISION

For additional information circle #123

DYNACORD DIGITAL DRUMS AVAILABLE IN U.S. THRU EUROPA TECHNOLOGY

The electronic drum kit, manufactured in West Germany by Dynacord Company, consists of seven 13-inch pads, and one 20-inch pad; plus The Percuter, an eight-channel digital drum computer. Each channel operates a plug-in EPROM cartridge that can be easily interchanged. All eight sounds can be driven dynamically by the drum pad or any other kind of triggering — microphones, sequencers, drum machines, or any other pulse source.

A library of over one hundred sounds including rock, fusion, and electronic drums, as well as special effects cartridges, is available. Other features include individual sensitivity controls, pans, stereo and individual outputs, and headphone output with separate mix.

Recommended retail price of the Percutter is $895; the 13-inch pads cost $150, and the 20-inch pads $230, cartridges sell for between $60 and $90.

EUROPA TECHNOLOGY INC.

For additional information circle #124

TOA INTRODUCES MODEL 380SE SPEAKER FOR ELECTRONIC MUSIC

Development of the new 380 SE is said to result from TOA's observation that a speaker meant only for "traditional" sounds cannot live up to the powerful levels and complex timbres of electronically-created, digital, direct-to-disk, and synthesized music. The new unit is described as a powerful three-way system that adds virtually no coloration to the original audio. The 380SE's configuration incorporates an exponential horn tweeter and Thiele-Small aligned bass reflex design (thus providing greater efficiency and a wider bass range).

The system provides continuous high power handling of 360 watts, and contains full-range inputs, bi-amp and tri-amp connectors, and four bridging connectors. Flush-mounted on the upper side of the speaker are

For additional information circle #125

New Products

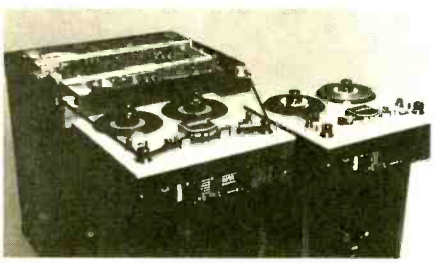
two level controls: one for mid- and one for high-frequencies, enabling output to be tailored to performance requirements and the room acoustics.

TOA ELECTRONICS INC.

For additional information circle #126

GAUSS INTRODUCES SERIES 2400 HIGH-SPEED TAPE DUPLICATOR

The new Series 2400 includes the capability to duplicate metal-particle and chromium-oxide cassettes, as well as microcassettes. System features include adjustable dual-capstan servo system; front-access modular electronics; an efficient tape loading system; unique hub locks; precision tape packer arms; replaceable tape cleaner cartridge; and can be utilized for either cassette, eight-track cartridge, microcassette, and ¼-inch reel-to-reel. The system consists of a master reproducer and endless tape loop bin, comprising the master playback system and slave recorders.



The duplicator can be easily interfaced to other Cetec Gauss high-speed duplicating systems, the company says, including its popular Series 1200, and includes a 10 MHz bias system and duplication ratio speeds of up to 128:1 to increase productivity and assure product quality.

"The Series 2400 introduces many innovative features, including new electronics, new systems and new mechanics," says Mort Fujii, company president. "It utilizes time-proven methods and concepts that expand on existing Gauss technology without sacrificing Gauss standards of performance and dependability."

CETEC CORPORATION

For additional information circle #127

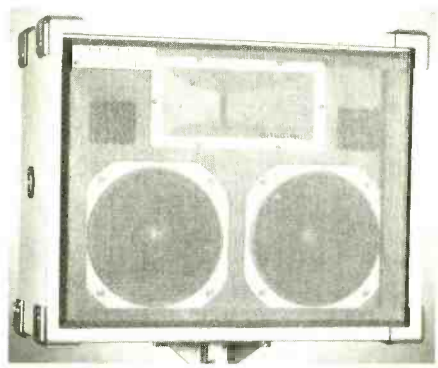
YAMAHA INTRODUCES S3208H SOUND REINFORCEMENT SPEAKER

The S3208H is a compact two-way sound reinforcement speaker system designed for a wide range of applications. Unlike many other compact speaker systems, the S3208H cabinet has been developed to accommodate a number of mounting applications — it can be stand-mounted, stacked, suspended, wall-mounted, or used as a slant-style foldback (stage) monitor.

For lower midrange and bass frequencies, the S3208H incorporates two high performance, eight-inch carbon-filter cone woofers.

The upper mid-range and HF horn is specifically designed to be flush mounted, for ease of shipment and packing into tight spaces. For ease of handling, a heavy-duty handle is placed in a rounded recess that allows balanced carrying with a minimum of wrist strain. For protection, signal jacks are deeply recessed into the rear of the cabinet.

Special interlocking corners allow the speakers to be easily and securely stacked on stage. In addition, four types of adaptors and brackets are available for a wide range of uses.



Wide dispersion and an unusually smooth upper midrange are said to make the S3208H excellent for main speakers, vocal monitors, and general music reproduction. Capable of handling up to 250 watts of continuous power, quoted frequency range is 65 Hz to 17 kHz. Suggested retail price of $545.

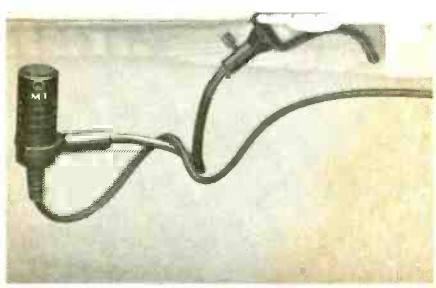
YAMAHA INTERNATIONAL CORPORATION

For additional information circle #128

FENDER INTRODUCES MINIATURE MICROPHONE SERIES FOR ON-STAGE DRUM AND GUITAR MIKING

The M-1 is a miniature condenser microphone that features a directional pickup pattern, flat frequency response, and is designed to handle 148 dB SPL without distorting.

The new microphone interfaces to mixing consoles via a pocket-size battery/electronics module that provides a switchable high-pass filter, as well as a notch filter tunable from 50 to 320 Hz to reduce feedback when miking acoustic instruments. The unit operates on internal battery, or 48V phantom power for additional dynamic range. Supplied accessories include tie clip, foam windscreen, and hard carrying case.



To help sound engineers take advantage of the M-1's multi-application potential, the company also offers three specialized mounting clip kits, made of black, vinyl-clad flexible wire, which can be cut to any desired length to form a "custom gooseneck." The general application A-kit includes a spring-loaded clamp for attachment to mike stands, piano lids, etc; the eyeglass/headset B-kit includes mike holder, flexible wire, and small padded clip, making it ideal for both singers and speakers; and the C-kit is designed for attachment to flat surfaces, such as acoustic guitar tops.

The M-1 carries a suggested professional net price of $175; application kits are priced from $19.95 to $49.95.

FENDER PRO-SOUND DIVISION

For additional information circle #129

CANARE INTRODUCES NEW LINE OF PROFESSIONAL CABLE REELS

The reels are constructed of tubular steel, with an E-shaped brace, and include heavy duty, permanently lubricated bearings. All models include a special three-position brake lever; in locked position, used during transportation, the reel will not rotate. In the soft-brake position, cable can be pulled from the reel, but friction prevents excess spillage when cable is pulled quickly. In the free position, the cable will pull from the reel easily and is ideal for rewinding.

For extra convenience, the R380 and R460 multichannel reels come with roll-around casters. Reels for single cables, which are smaller and lighter, are designed to be stackable, so they take up a minimum of storage space; several cables may be pulled at once while the reels are stacked.

Some cable reels include built in junction boxes with paired male and female XLRs per each channel of the snake, while others have a cutout and cable holders so the multipin end of the cable can be connected as desired. Multiple reels can be chained for very long cable runs.

CANARE CABLE INC.

For additional information circle #130

ALLEN & HEATH/BRENELL CMC 24 MIXER WITH COMPUTER-AIDED ROUTING SYSTEM

The new 24-input/16-bus console features a new three-band sweep equalizer, six auxiliary sends, and genuine solo-in-place on both input channels and tape monitors. It offers 16 full-function input channels and a further eight inputs, each having 16-buss routing, access to all six sends, and a two-band EQ section. Total format is 24 into 16 into 2, with 40-line inputs available for remix.

All input/output routing is handled by the console's microprocessor-controlled section. Complete routing status of the CMC can be stored as a "patch," and the on-board memory can retain up to 16 patches with full battery back-up. Mute status of channels and monitors can also be stored along with the routing information. A computer port can be linked to a microcomputer via an optional interface, to allow greatly increased patch storage, together with displays of channel routing and mute status and complex mute sequencing. The CMC 24 also provides the potential for sync-to-tape patch switching.

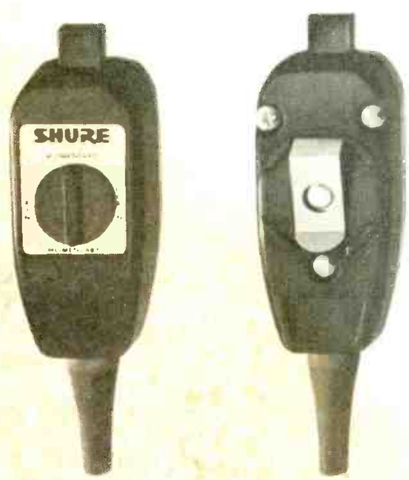
Designed for recording in situations where space is scarce, the CMC 24 provides input connectors that are accessible from the front and is the perfect solution for 'up-against-the-wall-recording'.

ALLEN & HEATH BRENELL (USA), LTD.

For additional information circle #131

SHURE DEBUTS A120SM IN-LINE ON/OFF SWITCH

A compact pushbutton switch, the A120SM can be used on virtually any type of microphone or electronic equipment, and is said to be particularly well-suited for use with lavalier and headset microphones. A cam mechanism built into the A120SM allows the user to instantly change it from a locking to a momentary, cough-button-type double-pole/double-throw switch.



The unit is constructed of durable black plastic, and is equipped with a belt clip that can be rotated in steps for maximum mounting flexibility.

SHURE BROTHERS INC.

For additional information circle #132

AUDIO KINETICS ANNOUNCES NEW FILM MACHINE INTERFACE FOR THE Q.LOCK SYNCHRONIZER

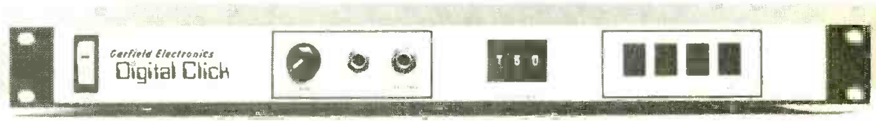
The new film-machine interface, Inverse Unifilm, allows film (projector or sound follower) transports to be used as the master machine without recording timecode on the stock, by taking the bi-phase pulses (or positive-tach direction) and generating a pseudo timecode within Q.Lock for the system synchronization. The new interface is supplied with a machine-control cable to interface to the machines' transport commands, so that full system control may be obtained from Q.Lock.

AUDIO KINETICS

For additional information circle #133

GARFIELD ELECTRONICS INTRODUCES "DIGITAL CLICK" METRONOME

The new digital metronome for film and video production provides clicks in both 24 fps calibration for the American film rate, 25 fps for Europe, and 30 fps to match the video frame rate. Tempos in all three calibrations can be specified with 1/32-frame resolution which, according to the company, allows greater flexibility by providing more calibrated tempos to choose from than ever before.



The Digital Click is said to produce click tracks of 10 times greated precision than previous metronomes with its 0.001% crystal based accuracy, and the enclosure occupies only one rack space. The unit features front and rear mounted jacks for external triggering and click output, and can drive headphones directly.

Recommended pro-user price of the Digital Click is $595.

GARFIELD ELECTRONICS

For additional information circle #134

New Products

NEW MC SERIES PA AND RECORDING CONSOLES FROM YAMAHA

The MC Series consists of three models —the MC1204, MC1604 and MC2404, — with 12, 16 and 24 input channels, respectively; other than input and meter configurations, all consoles have the same basic features. Each model has four group outputs assignable to two stereo outputs, plus two echo and two foldback sends.

The consoles are modular in construction, in blocks of four input channels for easy service when necessary. Each input channel features a pad switch and gain control with peak LED; three-band equalization with sweep midrange; two pre-EQ/pre-fader foldback sends; two post-EQ/post-fader echo sends; pan control; group select switches; and cue and channel on/off switches. All knobs are color-coded between input and output sections for easy identification in low-light situations, as well as making the visual signal flow easy to follow.

The front panel also includes a headphone/cue section and talkback input, each with a level control. The low profile meter panel shows output levels for groups 1 thru 4, foldbacks one and two, echo one and two, and left and right stereo outputs.

The back panel features low-impedance, electronically-balanced XLR and high-impedance ¼-inch inputs and channel insert in/out connectors for each input channel. Each module of four channels has a phantom on/off switch. Each of the four group outputs has an insert in/out jack, sub-mixer input and an electronically balanced XLR group output connector.

The MC1204 has a suggested retain price of $2,095, the MC1604 $2,695, and $3,795.

YAMAHA INTERNATIONAL CORPORATION

For additional information circle #137

NEW LEVEL AND ANGLE INDICATOR FROM PEIRCE-PHELPS PROVIDES DIRECT READINGS OF SPEAKER ALIGNMENT

Designed to set angles and/or measure them, the Inogon Level and Angle Indicator provides direct numerical readings of angles with a precision of 0.2 degrees. Pitch rise can be quickly converted to angles by a conversion table provided with the unit. The tool uses a new optical technology that is said to eliminate parallax errors. Angle measurements are based upon a change in visual patterns that occurs when light passes through two superimposed optical windows.

Physically, the new tool consists of an angle indicator calibrated in degrees that fits into a rule calibrated in inches. The indicator frame and window are made of high-impact plastic. To set angles, an adjustment screw on the indicator is turned to align the desired angle (in number of degrees) on a moving scale with a zero reference point on a fixed scale. Then the tool is tilted until parallel lines are seen in the window of the angle indicator, showing that the tool is now inclined at the desired angle.

If a number of speakers are to be set at the same angle, the tool can be used repeatedly without moving the adjustment screw, which can eliminate the time-consuming process of re-aligning an ordinary level and protractor for each speaker or horn.

To measure an angle, the device is placed on or against the surface of the speaker cabinet or mounting frame. The adjustment screw is rotated until parallel lines appear in the indicator window. The angle of the pitch or yaw is then read out on a scale in degrees.

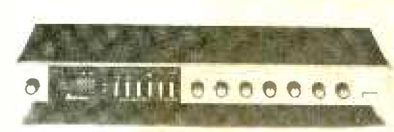
The Inogon Level and Angle Indicator is available in lengths of 10, 24, and 48 inches, magnetic or non-magnetic.

PEIRCE-PHELPS INC.

For additional information circle #138

IBANEZ INTRODUCES HD1500 PITCH SHIFTER AND DELAY UNIT

The HD1500 Harmonics/Delay and PC40 Preset Controller shifts the pitch of any material over one octave up and down, with a reduced processing time of 30 milliseconds and enhanced feedback capabilities.



The optional PC40 provides three presettable settings for real-time melodic harmonies, along with mode selection and effect bypass.

The HD1500 also provides a full-function digital delay with up to 504 milliseconds of delay. Included with the unit are a LED readout, "easy-touch" front panel switches, and multiple input/output configurations.

HOSHINO (USA), INC.

For additional information circle #139

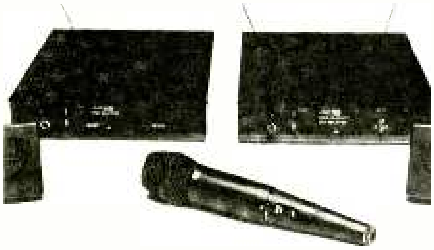
SAMSON INDRODUCES PHASE REFLEX HIGH BAND WIRELESS SYSTEMS

The new system's receiver features a Phase Reflex (true diversity) AB switching system, with helical filters, digital synthesized tuning and Phase Locked Loop tuning control to virtually eliminate drift and intereference. The true diversity Phase Reflex system is said to provide reception of up to 300 feet under adverse conditions, and up to 1,500 feet line of sight.

The receiver uses cavity construction for optimum shielding and reliability, and the microphone is built to house a variety of cartridges, with the Shure SM58 (dynamic) and SM85 (condensor) offered in initial production (others on order). The unit transmits on a crystal-controlled frequency in the 168 to 199 MHz range with an internal antenna.

Standard 9-volt alkaline batteries provide six to eight hours of continuous performance.

Other features include Auto-Squelch, which senses when the microphone is off and automatically mutes any noise in the receiver (patent pending); and diagnostic capability to scan the available frequencies for the "cleanest" RF channel.



The complete Phase Reflex system with an SM58 cartridge has a suggested list price of $1,295.

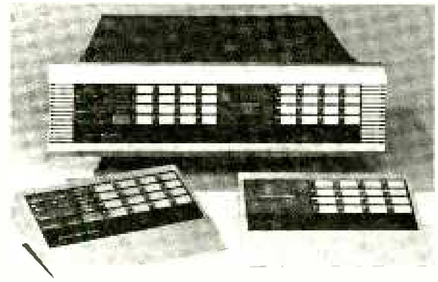
SAMSON MUSIC PRODUCTS

For additional information circle #140

INTEGRATED MEDIA SYSTEMS UNVEILS MODEL 200 ROUTING SYSTEM

The Model 200 Smart Switcher audio routing/mixing system can be configured in matrices of 32×8, 8×32, 16×16, etc. in a single 5¼-inch frame including power supply. Matrix configurations are in groups of eight, and can contain up to 256 crosspoints per frame. The matrix can be expanded up to 128×128, with up to 32 control stations.

The unit is available with both local and remote-control panels to access the matrix. It can be structured in multilevels (for example, stereo audio and timecode) and is easily field upgradable.



Features include four on-board memories for present matrixes; control via SMPTE RS-422 serial and RS-232 interface; external contact closure (GPIB); vertical-interval switching; party-line control; multiple-source summing capabilities; and optional VCA level control.

INTEGRATED MEDIA SYSTEMS

For additional information circle #141

DOLBY HX PRO SYSTEM NOW AVAILABLE FOR CETEC GAUSS CASSETTE DUPLICATORS

The Dolby HX Professional biasing device has been incorporated in the Cetec Gauss Series 2400 high-speed cassette duplicating systems, and is now being offered as a factory option conversion kit for existing Series 1200 high-speed duplicators.

HX Professional biasing technique is said to improve the high-frequency performance of cassettes with prerecorded programs that have high-energy HF content, and which can be recorded more accurately without sacrificing signal-to-noise ratio and distortion, performance.

According to Mort Fujii, president of Cetec Gauss, "HX Pro makes it easier to duplicate more accurate recordings, particularly of programs rich in highs and thus difficult to record on cassettes. HX Pro increases the useful dynamic range of tapes by improving its capabilities at high-level, high-frequency levels."

"By using the Gauss system, outfitted with the HX Pro, users have a wide tape selection (six pre-settable bias settings), varied duplicating speeds (64:1 and 32:1), and can significantly improve the audio quality of cassette recordings."

The HX Professional system is supplied with two preset positions for ferric-oxide and chromium-dioxide formulations. A third, adjustable position is provided for more exotic formulations, such as metal-particle tape. The speed control switch will automatically switch between 64:1 and 32:1 duplication ratio.

CETEC CORPORATION

For additional information circle #143

TIMELINK UNIVERSAL TIMECODE/SYNC DEVICE FROM AUDIO KINETICS

The new unit which can be used with various types of reference frequency, is designed to "get you out of trouble" whether the problem is poor quality timecode or an incorrect standard. TimeLink contains a timecode

New Products

reshaper, which will reconstitute poor code, plus a six-input "electronic gearbox" that can be set to produce an output reference frequency of a different standard from the input reference.

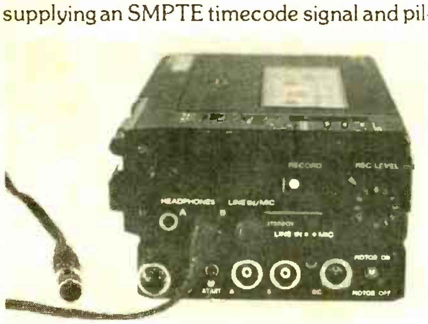
Inputs may be timecode, video syncs, FM, pulse, AC power frequency (internal), or internal crystal, and the output available as a field and frame-rate reference frequency, locked to the original. For example, a timecode input of 30 fps may be used to generate a reference that will enable a timecode generator (such as that contained within the Q.Lock Synchronizer) to lock to the original source, but generate, say, 25 fps Other applications include 50 Hz to 60 Hz reference conversion, or any ratio of the frequencies 48, 47.94, 49.94, 50, 59.94 and 60 Hz, which correspond to 24, 25 and 30 fps and their drop-frame equivalents.

AUDIO KINETICS

For additional information circle #145

AATON INTRODUCES WLD-6 MINIATURE TIME CODE GENERATOR

The new unit fits onto the Sony WM-D6 Professional Walkman cassette recorder, supplying an SMPTE timecode signal and pil-



otone on the right audio track, and mono recording on the left track. Once recorded, the on-location cassette may be transferred to either 16mm or 35mm mag fullcoat with an SMPTE cue track, or to multitrack tape.

Like Aaton's Clear Time Recording system, the WLD-6 may be initialized with a master clock, such as the "Origin C." It may also be initialized with an SMPTE generator, or by flicking the "start" switch when the counter is set at zero.

The coder offers a custom XLR-type microphone input that may be used in place of the WM-D6's minijack.

ZELLAN ENTERPRISES LTD.

For additional information circle #146

AUDIX ANNOUNCES UD-200 DYNAMIC MICROPHONE

The first of the UD-Series that incorporates a new air-suspension design, the UD-200 is a high output, low-impedance microphone with a quoted response from 50 Hz to 18 Hz, tight cardiod pickup pattern, and an integral acoustical pop filter.

The microphone, intended for on-stage and studio vocals, is available in black and non-reflective matte gray, and is supplied with holder and padded carrying pouch.

Suggested list price of the UD-200 is $179.

AUDIX CORPORATION OF AMERICA

For additional information circle #147

DOUBLE-FIFTEEN PA SPEAKER FROM JAI

The new D15AA double-fifteen Air Activator has a quoted usable frequency response from 50 Hz to 2 KHz, and is described as being ideal for several applications aside from PA low-end use; it can be used for drum monitor low-end, side fill low-end, keyboard low-end, and on-stage drum machine applications.

The cabinet measures 36 × 24 × 17 inches and is constructed from void-free 13-ply, ¾-inch baltic birch. Two styles of finish are offered: satin black laquer, or walnut stain. Two heavy metal grills, two sets of handles, and a locking ¼-inch connector are included. The D15AA is also available in a single configuration.

JOE'S AMALGAMATED INDUSTRIES

For additional information circle #148

TRIDENT LAUNCHES T.I.L. SERIES OF IN-LINE CONSOLES

While appearing to be of "in-line" construction — that is, without a separate monitor section — the T.I.L. console is said to avoid the confusing status changes normally associated with in-line mixers. This is achieved by the ability to route all or any combination of the eight auxiliary sends to either the input or monitor section. The result is described as an extremely flexible and simple to operate mixing console.

In addition, the user can not only route the entire equalizer section into the corresponding monitor section, but also the routing can be split so that either just the high and low or two swept midsections can be selected to monitor. As a result, during mixdown 60 line inputs all with equalization (66 including the echo returns) are available.

TRIDENT (USA), INC.

For additional information circle #149

APPLIED MICROSYSTEMS UNVEILS CM50 AUTO-LOCATOR

The CM50 is a microprocessor-based auto-locator that may be fitted to a large number of different multitrack machines, even ones that do not have a counter already fitted. The unit provides nine stored cue points, negative and positive offsets, shuttle between any two cues, and optimized slowing of the deck to a cue point. A time may be stored in a cue memory either via the numeric keypad or automatically from the current tape position, even in fast wind. The current tape time may be reset to zero or preset to any time selected on the keypad.

The unit comes complete with interface cable ready to fit to the tape machine; in the case of older machines a photo-sensor kit is also supplied. Transports that currently may be connected to the CM50 include Ampex MM1000, MM1100, MM1200 and ATR100-104; Fostex A8 and B16; MCI JH110, JH16 and JH24; Mitsubishi X-80; Otari MX-5050 series; Soundcraft SCM Series; Stellavox TD88; and Studer A80 MkI and MkII and B67; Tascam 48, 58 and 80-8; and 3M M79.

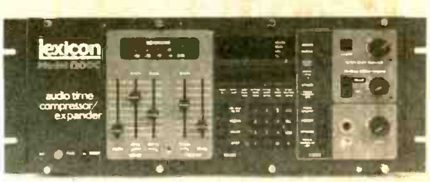
APPLIED MICROSYSTEMS LTD.

For additional information circle #150

LEXICON MODEL 1200C AUDIO TIME COMPRESSOR/ EXPANDER FOR VTR AND EDITORS

The new unit communicates with a variety of one-inch VTRs and editors via the Sony BVH-2000 protocol (RS422/232) to devices and systems that offer time compression/expansion editing software. The Ampex VTR interface is said to greatly improve servo lock-up time, reducing it to less than three seconds; as a result, the pre-roll requirement is dramatically reduced.

Timing capability is said to have been improved to better than one second per hour of play time. Improved input level matching and signal-to-noise characteristics are optimized for broadcast applications.



The Model 1200C is priced at $8,500; an optional RS422/232 serial data communications board, priced at $1,000, is required for the Sony BVH-2000 protocol feature. A field upgrade for earlier 1200 versions is available at a nominal cost.

LEXICON INC.

For additional information circle #151

FURMAN SOUND ANNOUNCES QN-4 NOISE GATE

Employing sophisticated variable pulse width modulation technology, the new four-channel unit is said to provide extremely low noise and superb frequency response specifications. The QN-4 features a fade-time control for each of its four independent channels, which allows the user to set the slope of the muting action — from a fast drop off to a gentle unobtrusive fade — to suit the program material.

The device also contains extremely wide-range threshold controls that enables it to function in a wide variety of applications and with almost any audio source.

The QN-4 four-channel noise gate carries a suggested list price of $395.

FURMAN SOUND INC.

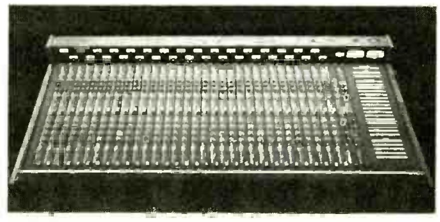
For additional information circle #152

. . . continued overleaf —

New Products

PULSAR INTRODUCES "ON-TRACK" RECORDING CONSOLE SERIES

The On Track Series is a semi-automated I/O console with eight VCAs (any channel can be a master), and each channel has a VCA defeat switch. Programmable mutes are standard, and a fully balanced patch bay also is included. A unique feature of the console is said to be an additional 40 patch points, allowing true stereo effects.



Other features include balanced in/outs, stereo monitor sends, multitrack monitoring, three-band sweep EQ, separate line/mike/tape trims, PFL, AFL and Tape solos, slate and talkback, calibration oscillator, phantom power, VU meters on all inputs, as well as L/R and solo.

Recommended retail price of a balanced 32-input On Track console is in the region of $26,000; 16-, 24- and 48-input mainframes are also available.

PULSAR LABORATORIES INC.

For additional information circle #154

OTARI RELEASES SMPTE/EBU TIMECODE IC

The new I-0055 is a CMOS LSI gate array configured to accept SMPTE/EBU longitudinal timecode as input, and produce both 8-bit parallel and serial data as output. The IC will accept buffered SMPTE/EBU LTC at speeds from 1/100 to 100 times play speed, allowing it to be used in wide-band synchronizer or timecode reader applications requiring easy CPU interface with no additional circuitry. It is user selectable to output either timecode or the user bits.

Features include single-ended 5-volt power supply and CMOS fabrication for lower power consumption; internal register over/underflow indicator outputs; tape direction output; data outputs from high-impedance tri-state drivers for easy data buss interface; and data outputs are asynchronous with the input timecode allowing data output at any rate.

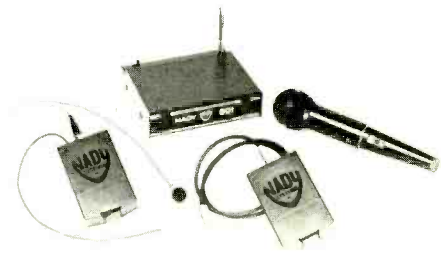
OTARI CORPORATION

For additional information circle #155

NADY SYSTEMS INTRODUCES 601/701 VHF WIRELESS MIKES

Designed to replace the 600/700 models, the new 601 is a single-channel receiver, the 701 is a true diversity VHF system. Three transmitters are available: handheld (601/701-HT), lavalier (601/701-LT), and instrument (601/701-GT)types.

Improved filter circuitry is said to allow closer channel spacing and larger multichannel capability. For maximum flexibility, the new units include a switchable balanced line and mike output, an unbalanced line output, and a monitor output to drive a headset or for line feed for instrument amplification. LED trees on the receivers indicate signal strength and audio level. Dynamic range is a quoted 120 dB max SPL to A-weighted noise level.



Body pack for the lavalier and instrument transmitters measures only 3¼ inches high, and clips on to a belt or slips into a pocket.

The 601/701 hand-held microphone/transmitter is a tapered, compact design with no protuding antenna; elements available include Shure SM-58 and SM-85, Electro-Voice EV-76 and BK-1, and Audio-Technica ATM-91 and ATM-41.

NADY SYSTEMS INC.

For additional information circle #156

CETEC VEGA MODEL 66A PORTABLE WIRELESS MIKE RECEIVER

The new battery-powered wireless-microphone receiver features lower power consumption than the previous version, for extended battery life, and can operate from an external DC power source or +10.5 to +18 VDC, or from its internal battery pack (four, 9-volt alkaline cells).

Fully compatible with all previous PRO transmitters, with the optional Dynex II audio processor, the Model 66A is also fully compatible with, the company's new T-Series transmitters. The receiver operates on any crystal-controlled frequency between 150 and 216 MHz; the preselector is described as a true two-pole, helical-resonator filter. The use of a combination of LC and multiple-pole crystal IF filtering is said to provide outstanding IF selectivity and adjacent-channel rejection. The wideband, FM demodulator has low distortion (system THD is typically 0.3% or less) and excellent dynamic range. With the Dynex II processor, usable dynamic range is quoted to be in excess of 100 dB.

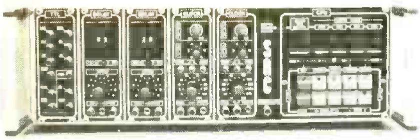
CETEC VEGA

For additional information circle #157

TIMECODE INTERFACE FROM EUROPA TECHNOLOGY

The SMPTE Reading Clock, manufactured by Friend-Chip Company of West Germany, is designed to solve problems that can occur when trying to synchronize synthesizers, sequencers, and drum machines to SMPTE timecode. The SRC reads and generates 30-frame or drop-frame timecode. A start time is entered that informs the unit's CPU of the proper time to begin the synchronized instrumentation. A built-in metronome function ensures that all instruments are locked into the same beat.

Independent 63 millisecond delays are incorporated to permit the time-shifting of various instruments relative to the start time, and tempo is adjustable from 30 to 255 beats per minute.



New software updates available by September will allow for programming up to 32 cue points, as well as programmable tempo changes. Also available in the Fall will be an input module designed to sync all instrumentation to any click or trigger input.

EUROPA TECHNOLOGY INC.

For additional information circle #159

AUDIOTEC UNVEILS ACE 36/24 CONSOLE

The unit if fully modular with a maximum of 36 inputs and 12 stereo outputs (for 24-track recording, and offers the following features per channel: 48-volt phantom powering; phase reverse; mike/line switching; tape return; Hi-Cut/Low-Cut; four-band EQ; (parametric on Mid #1 and #2 with switchable bandwidth); four cues switchable, pre/post; mute; solo; and peak-reading and low-level (–20 dB) LEDs.

This unit was recently featured at May AES and is expected to be available through a select number of quality dealers soon.

AUDIOTEC

For additional information circle #160

TAC SCORPION CONSOLE FROM AMEK

The new low-cost, high-performance console is derived from the TAC 16/8/2 system, which the Scorpion replaces. Principle features include fully modular construction in a welded steel frame; 16 routing busses, plus a separate stereo bus; four-band EQ with

swept mids and selectable turnover points; four aux sends; two assignable aux returns; fader reverse function; metering with switchable peak/VU ballistics; plus stepped chassis with horizontal fader section, 10-degree module angle, and penthouse meter hood.

The Scorpion is available in two frame sizes: a 27- and a 43-position. A 16/8/2 with eight-track monitor is a short frame will retail at $5,950; the same configuration shortloaded into a large frame will carry a retail price of $6,950.

AMEK CONSOLES INC.

For additional information circle #161

SOUNDTRACS UNVEILS M SERIES CONSOLES FOR RECORDING AND LIVE SOUND

The new Series is a modular console designed for eight-track studio, sound reinforcement, theater-sound applications. Each input module provides four-band equalization; six auxiliary sends, switchable pre/post; plus eight buss and master routing.

The output section provides switchable EQ for either group sends or tape returns; tape returns to group and masters; stereo return to masters; six auxiliary sends switchable from group to a tape return; and, for most theater applications or studio monitor, an inclusive four-way matrix for each group. Monitoring is provided by a 10-element VU meter bridge with illuminated legend.

Packaged in a rigid steel frame, utilizing aluminum extrusions, the M Series is available as a 24-into-8 or 32-into-8, with any permutation using blanks.

Availability is November, with a target price for the 24/8 of approximately $8,500.

MCI, MUSIC

For additional information circle #162

— continued from page 36 . . .

News

SPARS SETS UP TELECOM NETWORK FOR STUDIO MEMBERS

Working in cooperation with IMC Communications, the Society of Audio Recording Studios currently is establishing a telecommunications network linking its members. The network will provide a quick and efficient means for member studios to exchange information regarding technical problems and solutions, equipment needed or for sale, replacement parts needed, rental needs and other information that will assist in studio operation and management. In support of the network, Digital Entertainment Corporation has provided SPARS with a Leading Edge personal computer plus auxiliary equipment and programming.

Installed at SPARS's new location at United Western Studios in Hollywood, the Leading Edge computer, complete with word-processing software, Hayes 120B modem, and Crosstalk TC software, will serve as an office management system, and also produce an Electronic Newsletter that goes on-line this fall. The new SPARS On-Line Service will contain up-to-the-minute rental company information and pro-audio bulletin board for immediate exchange of information among members.

The On-Line Service is distributed electronically via the IMC International Communications Network, which is said to be specifically geared to the needs of the entertainment industry. Besides the distribution of its Newsletter, the Society intends to use the TC network to develop applications software for tape ordering, maintain a stolen equipment "Hot-Line," and disseminate equipment and technical information.

Also, David Teig, former SPARS eastern coordinator, has been named national coordinator for regional meetings, a position that has been established to provide staff assistance to members in the organization and execu-

Some 100 executives representing all facets of the audio cassette duplicating industry participated in a recent technical seminar specifically devoted to the quality of pre-recorded cassettes. According to its organizers, Electro Sound, Inc., the seminar — held in Sunnyvale, California from August 22 thru 24 — exceeded all expectations. "It was necessary to turn away many interested people who would have otherwise attended if we had originally arranged sufficient space" said Bob Barone, president of Electro Sound. Seminar co-sponsors included Agfa-Geveart, Athenia, BASF, Capitol Magnetic Products, Columbia Magnetic Tape, Digital Entertainment Company, Dolby Laboratories, E.I. Dupont, Hercules, ICM, IPS, JRF Magnetic Sciences, Pfizer, Saki Magnetics, Shape and Studer Revox America.

According to David Bowman, senior VP of Electro Sound, "It appears that the only way to significantly improve the quality of the mass-produced, pre-recorded cassette is to establish a common understanding of the limitations of both the process and the medium, and for all concerned to optimize their individual contributions to achieve the best possible result."

Comments made by many participants confirm that such a seminar was a long-awaited and necessary part of the cassette industry. For those who did not attend, Electro Sound will make available, at no charge, printed transcripts of the presentations; contact the company's Customer Service department, 160 San Gabriel Drive, Sunnyvale, CA 94086. (408) 245-6600.

tion of regional meetings. The board also has approved the development of a group health insurance program. Initial specifications have been established and the program was scheduled to be available to studio members by early October.

SONY REPORTS MAJOR SALES SUCCESS OF DIGITAL HARDWARE IN EUROPE

During the past several months, Sony Broadcast's Pro Audio Department — the division responsible for the marketing and sales of digital systems in Europe — reports major sales to **Audio Fx**, London, which purchased a PCM-3324 DASH digital multitrack with RM-3310 remote control unit to be included in the company's range of equipment for hire; **Sarm West**, London, which purchased a PCM-3324 and RM-3310; producer **Steve Levine**, already the owner of a PCM-3324, and who ordered an RM-3310 remote control unit to use with it (see "Production Viewpoint" profile in August issue); and **EMI Abbey Road**, which is extending its digital facilities and has ordered another complete PCM-1610 editing system. Further PCM-1610 systems have also been ordered by consultant **Tony Faulkner**, and **Tape One** which is adding a second system to its installation.

Recent PCM-1610 system sales have taken the total sold worldwide to over 300. Among U.K. sales were Townhouse Studios, Angel Studios, CBS, Master-Room, Steve Levine, Abbey Road, Nimbus, Hugh Padgham, CTS, Fisher Lane and HHB. In Europe, Sony Broadcast have secured sales of five PCM-1610s to **Hungaroton**, Budapest, to extend its existing equipment; a further PCM-1610 to **Supraphon**, Czechoslovakia; and a complete PCM-1610 editing system to **Opus**, Bratislava. **EMI Music** has also purchased a number of PCM-1610s and support equipment for use in many of its overseas facilities, and the Hayes Tape Duplication Plant.

In Germany sales are reported to continue at a brisk rate, and several systems have been installed in France. Companies in Belgium, Holland, Italy, Spain, Sweden and Switzerland have also purchased PCM-1610 systems.

In total, eight PCM-3324 DASH multitracks now have been sold in Europe; five are in the U.K., and one each in Italy, Holland and Germany.

AUDIO COURSE OFFERED AT LINCOLN INSTITUTE

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. . . continued on page 211 —

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MISCELLANEOUS

News

— *continued from page 207 . . .*

thus providing the company with additional working capital.

According to CompuSonics president David Schwartz, a comprehensive marketing agreement between the two companies will be forthcoming by the end of November. In addition to those points already outlined above, the agreement will contain an option, based on Nissho's performance in selling the DSP-2000 systems, granting Nissho exclusive rights to export to Japan and/or license the DSP-1000 for production in Japan.

EV PREDICTS INCREASING SALES OPPORTUNITY IN CHINA

"Sweeping governmental changes in the Peoples Republic of China are beginning to stimulate great activity in improved sound systems," according to Jim Long, director of marketing at Electro-Voice. Long and Don Krajecki, EV director of export marketing, recently visited the Peoples Republic of China while on a speaking tour of the Far East and Australia. Long presented day-long seminars in China, Thailand, Japan, Singapore and Taiwan focusing on two topics: "A Selective Review of Speaker Components and Their Applications in the United States;" and "Microphones — a Discussion of Basic Types with Emphasis on Performance Differences and Application Guidelines."

"Even though pro-sound activity is not new in the Far East, the actual use and application of pro-sound equipment is not as prevalent as it is in the United States," Long says. "For this reason, the hands-on demonstrations and detailed examples of working installations were a popular part of our presentation on professional sound reinforcement products.

"The China presentation was particularly interesting," he comments, "because of a phenomenal interest in pro sound. I was impressed by the participants awareness of technical papers and books published in the United States." While in China, Long addressed the largest group of the tour, some 350 performers, professors and sound professionals from all over the country.

EV has expanded an already strong distributor/customer base in the Far East to include a new representative in China itself, according to Long. In cooperation with the Guangdong Provincial Performing Arts Company, EV's Hong Kong distributor is offering ElectroVoice equipment in a new service center located in the Chinese city of Guangzhou, formerly Canton. The service center formally opened in mid-April, encouraging Long's visit to the Orient. "In this city of five million people, the center will serve the ongoing audio needs of Chinese musicians and performers," Long predicts. While EV also has company-owned facilities in Japan, Australia, Switzerland and Canada, it is an unusual and unique situation for a western company to be represented in China, according to Long.

ALTEC LANSING COMPLETES CONSOLIDATION TO OKLAHOMA CITY HEADQUARTERS

The relocation of the company's administration, engineering, marketing, sales, data processing, and literature departments completes the process of relocation begun in 1982. During that year all loudspeaker manufacturing was consolidated in the larger Oklahoma City facility, an operation that was followed in 1983 by the West Coast warehouse, horn and sheet metal fabrication, and electronics assembly. The Oklahoma City plant was originally occupied in 1963 by Altec's University Sound Division.

The address for the Oklahoma facility is 10500 West Reno Avenue, Oklahoma City, OK 73126. (405) 324-5311.

BASF SEES BRIGHT FUTURE FOR CASSETTE WITH PROFESSIONAL RECORDING MOVING INCREASINGLY TO DIGITAL

Wolfgang Wiegel, audio product manager for BASF, Germany, forsees a long future for the Compact Cassette as an "even clearer choice among 'mobile' sound carriers," although the technology probably will move rapidly toward digital cassette recording techniques.

Speaking before the New York Chapter of the AES in June, Wiegel also predicted a rapid shift on the professional side toward digital technology, with magnetic tape continuing to dominate home video, and metal-evaporated pro-

News

ducts most important in data storage.

During his address, Wiegel traced the history of magnetic recording from its beginnings in 1888, when Oberlin Smith sketched out its basic principles, to the latest developments in digital and metal-evaporated technology. The occasion was the celebration of the 50th anniversary of magnetic tape, developed and introduced by BASF in 1934.

Other speakers on the AES program included Peter Hammar, curator of the Ampex Museum of Magnetic Recording, who concentrated on the hardware development; Dr. William Lafferty of Wright State University in Dayton, Ohio, who discussed the technical development of magnetic recording; and Stephen Temmer, president of Gotham Audio Corporation, New York, who provided an overview of studio recording and broadcasting in the U.S.

Wiegel recalled that as early as 1932, an agreement was reached between two German companies, AEG and BASF, to develop magnetic recording technology; AEG was to work on the hardware, and BASF on the software.

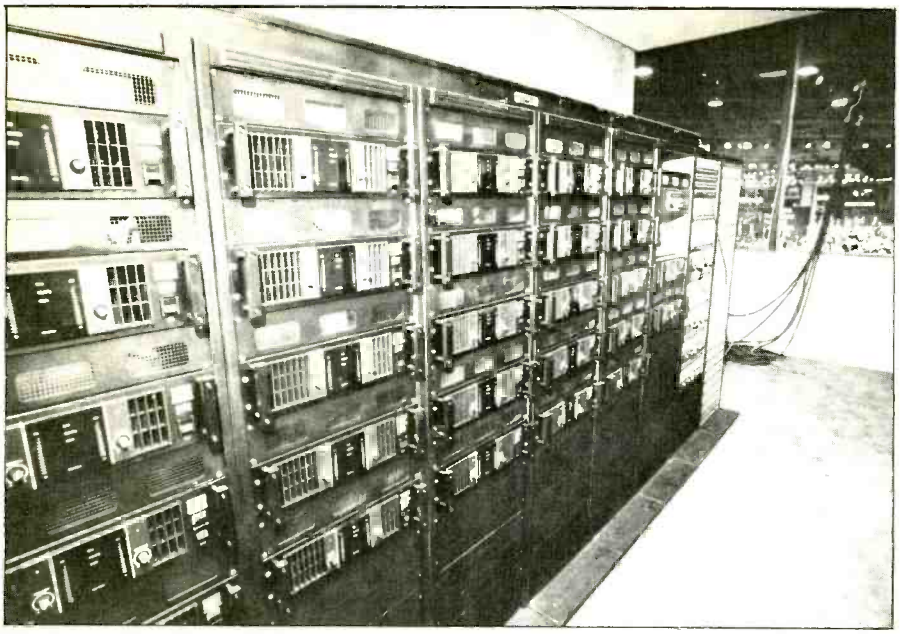
"The first experiments took place during the last days of 1932, and BASF was already on the right track," he says. "A 20-micron thick top coat containing carbonyl iron mixed with cellulose acetate as a binder was applied to a 30-micron thick cellulose acetate base. This was the first genuine, coated tape."

By June, 1934, Wiegel continued, AEG and BASF agreed that the system, now called "Magnetophon," was ready for introduction, but AEG called off the launch because of a persistent buzzing noise in the tape drive and amplifier units. "The next suitable date for the debut was the Great German Radio Exhibition of 1935, held as usual in Berlin," he explains. "The spectacular success of the Magnetophon at the Radio Exhibition made the prospects bright enough that BASF placed an order for a production machine to manufacture tape in larger quantities."

In the summer of 1936, BASF switched to a new formulation, ferric oxide, which had higher coercivity and retentivity, and this tape passed its first practical test in November of that year when BASF engineers recorded a concert given by Sir Thomas Beecham and the London Philharmonic in BASF's concert hall. By 1938 the system was in use for German radio broadcasts.

In conclusion, Wiegel says that the ensuing half-century have been years of steady progress. "We are convinced that this technology will remain every bit as important, though the forms it takes and the carriers used may undergo substantial changes."

During the recent premiere of *Metropolis*, the audience was treated — possibly for the first time during a public screening — to theatrical playback of first-generation digital audio . In the projection room at the Academy of Motion Picture Arts & Sciences' Samuel Goldwyn Theatre in Beverly Hills, two Sony PCM-3324 digital multitracks — one for each film projector — were interlocked via a SMPTE timecode track to picture during replay of Georgio Moroder's new musical score. Pictured here, from left to right: Steve Boze and Moroder's technical director Dave Concors, with Scott Spector, West Coast sales engineer for Sony Pro Audio, and Rick Plushner, Sony's national digital sales manager.



The recent Republican National Convention marked the first use of Dallas Convention Center's new sound system. Specified for the City of Dallas by Stan Miller of Stanal Sound, and installed by Audio Technical Services of Vienna, Virginia, the system's 407 speakers are driven by 52 Fender 2244 power amplifiers. Mike Abbot, assistnt to Miller, who served as the Republican National Committee's sound consultant, reported that the new amplication package stood up well. "The convention was an extremely demanding application. The amps were running in the four ohms bridged mode and were being required to produce in excess of 1300 watts each. We averaged 101 dB on the convention floor and still had 20 dB of headroom. In all regards, the Fender amps performed flawlessly, and lived up to their fully professional specifications."

AVC SYSTEMS BUSY ON RECENT PROJECTS

The Minneapolis-based equipment supply company, which now is located in the complex formerly occupied by Sound 80 Recording Studios, recently was contracted to supply and install a sound system for the new **Ordway Music Theatre** in St. Paul, Minnesota. Also, the **Carlton Celebrity Room** in Bloomington, Minnesota, has just finished installation of a new flown JBL loudspeaker cluster designed by AVC.

The firm has completed installation of a 24-track recording studio for **Jesse Johnson**; an AV conference room for **Fallon, McElligott and Rice**, and **Kolesar and Hartwell**, two well known advertising agencies; and a corporate production/presentation room for **Deluxe Check Printers, Inc.**, of Shoreview, Minnesota.

— News Notes —

Audio Kinetics' MasterMix console automation systems have been installed in studios in be Sweden, USA, UK, with additional orders from Finland, Holland, Italy and Australia. **Polar Music**, Stockholm, recently took delivery of a MasterMix system, for use on its new Calrec console (see "Studio Update," August 1984 issue, page 98), while Surrey Sound, Olympic Studios and Marcus Music in England are additional MasterMix installations on Harrison MR4, Raindirk (a retrofit including VCA Faders) and Harrison MR3 consoles, respectively. The company also reports that well over 650 Q.Lock synchronizer systems have been installed world-wide . . . **Uher Werke Munchen**, the West German manufacture of cassette and reel-to-reel tape machines, has set up a new U.S. office to handle sales and distribution. In addition, a comprehensive parts supply and factory-trained technicians will be available for service and repair of all existing Uher products. Designed primarily for remote or portable recording, models are available for film and broadcast production, and transcription and documentation work. The new company address is: Uher of America, 7067 Vineland Avenue, North Hollywood, CA 91605. (818) 764-1120. ■■■

— People on the Move —

• **James A. Panzer** has been promoted to the position of senior consultant at **Peter George Associates**, where he will continue specializing in performance facilities planning and systems design. In addition, Panzer will now be involved with communications facilities and systems design. Also, **David Harvey**, formerly of Imero Fiorentino Associates, has joined the company as a consultant, and will be involved in the design of lighting and audio-visual systems.

• **Thomas E. Mintner** has been appointed vice president and general manager of **Studer Revox America, Inc.** Minter has served as director of Studer Products at the company since 1982, and he will continue to direct all Studer Division activities in his new position. Concurrent with his own advancement, Mintner announced the promotion of **Doug Beard** to the post of director of technical and marketing services for the Studer Division. Formerly national service manager, Beard will now take a more active role in mark[illegible]nd key customer relations. Some of Bea[illegible]ice-related duties will now be handled b[illegible]in Knox**, newly promoted to service and quality control manager for the Studer Division. **Hans D. Batschelet**, who served as company president of Studer

News

Revox America from early 1983 to mid-summer of this year, has left the firm to return to his native Switzerland.

• **Joey Newman**, is **Fairlight Instruments US'** newly appointed executive director. In his new capacity, Newman currently is developing three new interfaces for the CMI: the Model VT-5 Voice Tracker; a SynthAxe guitar-interface; and MIDI-and SMPTE-compatible cards, to extend applications in the video and TV production areas. Also, **Todd Rundgren** has been named as a participant in the newly instigated endorsement program for the CMI. Over the next 18 months, Rundgren will be holding seminars and clinics in major U.S. cities.

• **Scott Schuman** has been named director of market development at **Dolby Laboratories**, where his responsibilities will include marketing and promotional support for licensing and engineering for the company. Some of the areas he is presently involved with include Dolby HX Pro "headroom extension," C-tape software, noise reduction for VHS pre-recorded videotapes, and the company's Digital Audio System. An expanded program to extend communications with the U.S.-based marketing arms of their overseas licensees will come under his direction.

• **Richard Lee** has been named national product and systems manager for **Sony Professional Audio Division**, where he will be responsible for pro-audio product and systems planning, and will interface with Sony factories in the U.S. and Japan. Most recently, Lee was vice-president/general manager of Criteria Recording, Miami, and previously spent eight years in the related fields of acoustics consulting, audio/video/film systems design, maintenance engineering and technical facilities executive management. Also, **Scott Spector** has been named west-coast engineering manager-digital audio for the same division, and will provide comprehensive technical support for Sony digital audio products. He previously held the position of technical supervisor with Sony's Video Communications Division, and prior to joining the company served as senior support engineering manager for MCA Discovision's manufacturing facility. On the sales front **Andrew S. Munitz** has been appointed eastern regional manager to coordinate sales of Sony Pro-Audio products to broadcast and OEM markets. Most recently, Munitz was western regional sales manager for BGW Systems, and has also been involved in recording studio design and construction, sound engineering and film production.

•**Jeff Evans** and **Geoffrey M. Langdon** have been named as new regional sales managers for **Rupert Neve, Inc**, with Evans covering the western half of the U.S. from the company's Los Angeles office, and Langdon the east. In addition to selling consoles to recording studios, video production and broadcast houses, he will help clients to create customized boards to suit their individual needs. Previous positions include design engineer for Dolby Laboratories, and sales manager at Sony. Langdon will be based at Neve's U.S. headquarters in Bethel, Connecticut, and joins the company from Sennheiser Electronic Corporation, where he was vice president of engineering; he is probably best known, however, for his nearly 10-year association with AKG Acoustics, where he was technical manager.

•**Michael Uhl** has been appointed national sales manager at **Auditronics, Inc.** Formerly with Pacific Recorders & Engineering, Uhl will be responsible for coordination of all domestic sales activities, including the establishment of a direct sales program in selected areas of the country. ■■■

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LATE NEWS

SOUNDCRAFT REACHES AGREEMENT WITH GML TO DESIGN AUTOMATION SYSTEM

George Massenburg Labs has been contracted to design, test, and implement a new hard-disk-based automation system for present and future Soundcraft consoles. The new system will use features of the current GML Moving Fader Automation System, including real-time software, standard Motorola 68000-family microprocessors, and UNIX-like operating systems; large global RAM for fast, real-time list manipulation; the flexible, extendable GML decision list and complimentary I/O format; and a convenient, multilingual, user-definable operator interface.

The present GML automation systems, which has been in commercial use for over a year, is described as being the fastest and most accurate available. The Soundcraft system will be made available in several configurations starting with a compatible VCA system; all current releases are designed to be upgradable to a moving-fader design.

Soundcraft will also offer integral to its larger console series a proprietary tape-based design for those applications not requiring disk-storage media. As a third option, Soundcraft also offers the Audio Kinetics MasterMix automation system.

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Here's the microphone soundmen everywhere are talking about. And we're not surprised–we worked closely with top sound engineers to perfect our new SM87. It's a studio-quality supercardioid condenser mic with Shure's legendary road mic ruggedness.

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Natural sound all across the board. The extremely smooth response characteristics of the SM87 offer soundmen tremendous flexibility at the mixing board. Its vocal contoured response permits quick, easy equalization (many engineers think it needs no equalization). The SM87 provides incredibly accurate voice reproduction across the entire frequency spectrum.

A workhorse that handles like a dream. The Crowd Pleaser performs smoothly when other mics get the shakes. The exclusive Shure elastomer "space frame" shock mount isolates the capsule from virtually all hand-held and mechanical vibrations. And the built-in multi-stage filter minimizes popping and wind noise.

Built for a world of hard knocks. The SM87 will withstand years of use (and abuse) because it's "tough tested" to meet Shure's worldwide reputation for ruggedness and reliability. What's more, the SM87 far exceeds normal specs for resistance to temperature extremes and humidity. Ask someone who owns a Shure mic and you'll know what we mean.

Get on the bandwagon. The coming decades belong to the SM87. To find out why, call or write for our free brochure. Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204, (312) 866-2553.

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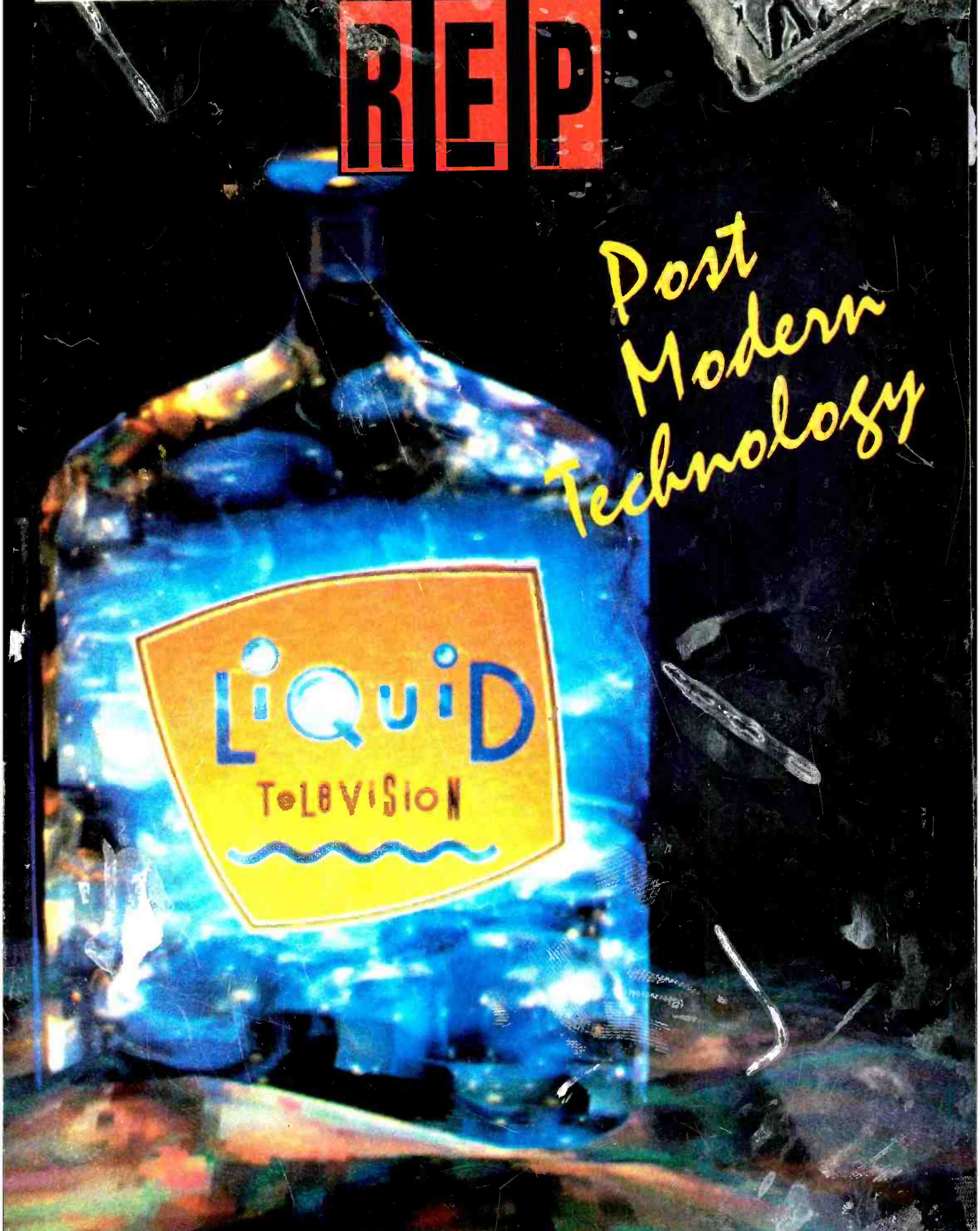
REP



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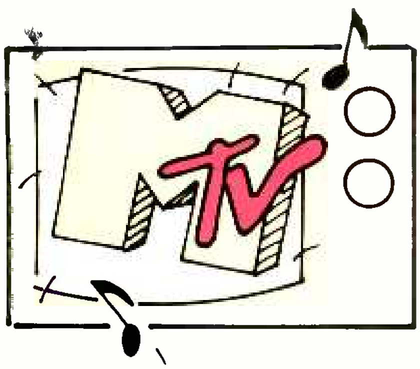

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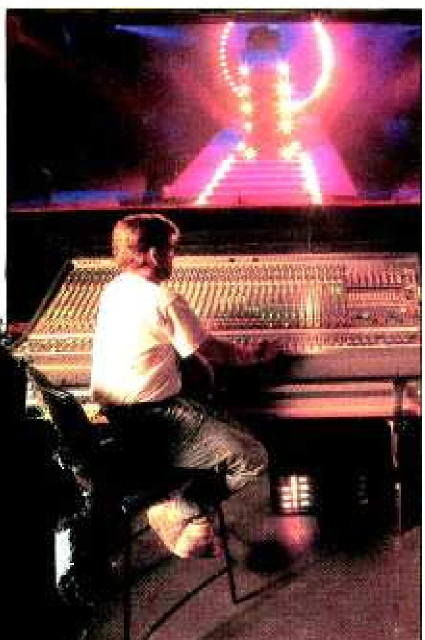

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On The Cover: Liquid TV logo courtesy of Keith Hatschek at Music Annex. Photo by XAOS, Inc.

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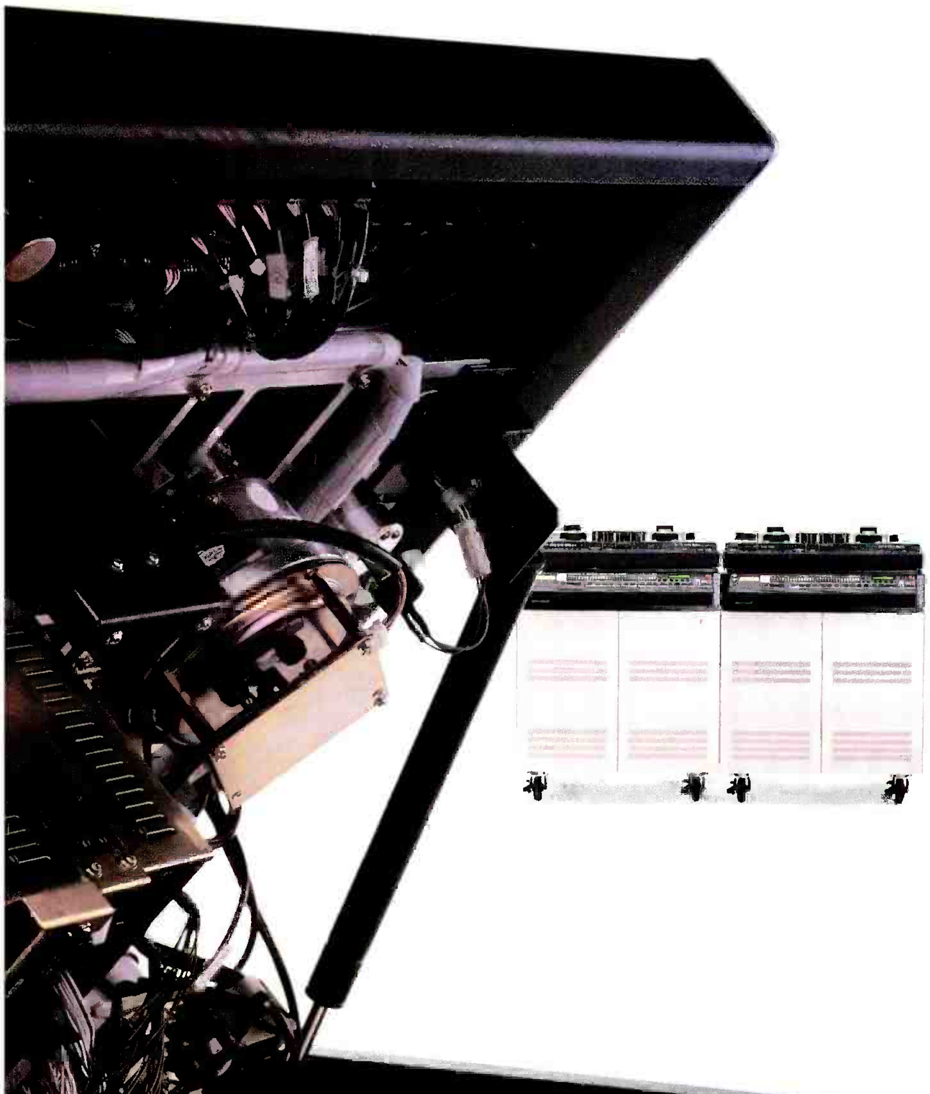
THE DTR-900II, MEANS BUSINESS, INSIDE AND OUT.



When a digital machine has earned the reputation of being " the best sounding tape recorder in the world," and at the same time delivers Otari's legendary dependability, it can only mean more satisfied clients, and more business for you. However, DTR-900 users tell us there are other reasons to own one that are just as compelling.

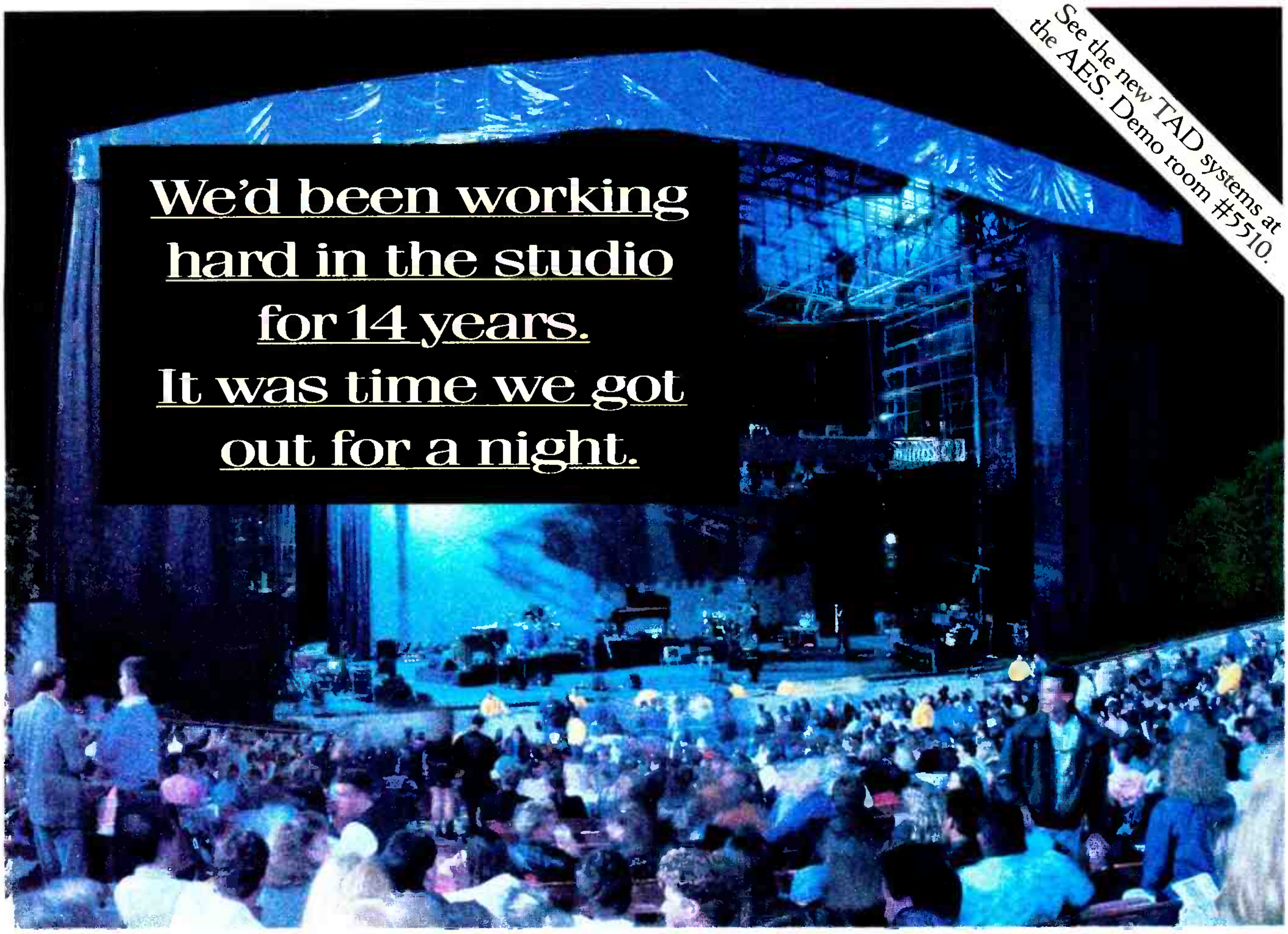
First, you're not locked into a single 48-track machine. Otari's 64-track system can be split for use in two simultaneous 32-track digital sessions. (One practical studio owner says he "uses one and rents one" when he doesn't need all 64-tracks!)

Another owner appreciates what he calls, "the best autolocator–it makes production easy by doing most of the work for you." (He also likes the "900's" unique ability for simultaneous recording on digital and analog inputs.) Still another says, "The DTR-900II's adjustable time advance feature is a lifesaver when we're using an outboard digital console."

Obviously, the DTR-900 means business for these owners. Call Otari at 415/341-5900 for more information about what it can mean to you.

OTARI®

Circle (4) on Rapid Facts Card



Spending years on end cooped up in small, dark rooms with a bunch of engineers takes certain special qualities. Durability, for one. We've always been known for that. Of course, clear, uncolored sound quality doesn't hurt, either. Or hand-assembled components, with gap precision to plus or minus one-millionth of an inch.

These features got TAD speakers into studios like Record Plant, NOMIS and Masterfonics. And the same features are now getting us out of them.

See, we had this funny idea that if TAD could make music sound terrific in a small room, we could make music sound terrific in a huge arena. And every outing we've had with Maryland Sound has proved us right.

Not that we won't still work our woofers off in studios from L.A. to London all day. But, at night, we'd like to get out and jam more often.

TAD Technical Audio Devices®
Pioneer Professional Products Division

Circle (5) on Rapid Facts Card

R•E•P is an applications-based publication targeted at professional individuals and companies active in the commercial business of studio and field recording, audio for video, live sound production and related fields. Editorial content includes descriptions and demonstrations of audio production techniques, new products, equipment application, maintenance and audio environment design.

Member, Business Publications Audit of Circulation
Sustaining member, Audio Engineering Society
Sustaining member of Acoustical Society of America
Member, International Communications Industries Association
Associate member of Society of Professional Audio Recording Services

From the Top

Who Do You Trust?

Here we are, living in a world where everything and every product is the best, has the most features and does the greatest job. Everything is great and performs perfectly, no matter the price. Just read the ads. Look at the fluffy articles and so-called product reviews that portend to expose the truth. The real truth is that it's common today for most magazines to be less interested in presenting reality to their reading public than they are in providing soothing words to the folks who buy ad space. My message today? Caveat Emptor. These are not honest times.

Let's talk about product reviews. Most other pro-audio magazines present themselves as if they were dealing in unbiased fact, but the truth is, many times they stop short of presenting the serious technical information that the rest of us pro's need. I don't know why. Real pro's are not scared of calculators. We can read graphs and understand measurements. Why do other mags chop out the tech and substitute fluff? Do they think we're all beginners or students? Or are they writing to the level of some least common denominator, some imagined unknowledgable reader? If I had a dime for every writer who came to me with a horror story about all the "good stuff" that another magazine chopped out of his piece, well ...

It's curious, but in the consumer audio, automobile, motorcycle and computer industries, to name a few, magazine reviewers really do take hard, detailed looks at products and compare them in mano-y-mano shootouts to competitive units. On specific features and performance merit, some units win and some lose. Even the losing products' manufacturers realize that the review's readers trust the magazine's writers, and that all the manufacturers are playing on a level playing field with their competition in regards to review ethics. If a product isn't as good, for whatever reasons, as a competitive piece, then it will fail in the marketplace. In effect, the reviewer is doing the manufacturer a service in the long run — it is showing him where and how to improve the product, and indicating (at no charge) what level of quality people will accept in a given price range.

In many industries, editors get notes of thanks from a given reviewed product's manufacturer, thanking them for pointing out the strengths and weaknesses saying, "We're going to improve that aspect you pointed out and we'd be honored to have you review it again." Mags often do. Usually it leads to a noticable improvement in sales. The readers respect the manufacturer for its honesty, and sales improve based on the large amount that they now know about the product, which contributes to familiarity and acceptance.

But in our industry? Usually not! If certain manufacturers get a truthful review, negatives and all, they stamp their feet and make nasty phone calls. Then they pull their ads. For most magazines, the fear of ruffling some feathers and losing ads is enough to kill any thought of conducting serious technical reviews. The readers lose in the end.

To their credit, quite a few manufacturers understand and support the concept of honesty. They are the first to say things to us like: "Great product review! As you pointed out, our $60,000 board (for example) does have 'X' crosstalk between these buses. To get another 6dB separation would have added two grand to the cost and required a whole different, more expensive design. It was a compromise we made to keep to the price point. And as it stands, it is still a wonderful value, with superior performance for the dollar. Remember, perfection costs money. We manufacture that too, but it's $200,000." We applaud the integrity of companies like that. They do the industry justice.

In the past several months, certain manufacturers who have had their products honestly and objectively reviewed in our pages have reacted loudly and rashly. Our response? We were sincerely surprised. In our reviews, we discuss things that anyone having access to an Audio Precision stack or a Techron TEF system can measure themselves, in real-world environments. We try to verify our measurements with manufacturer's staff engineers before the articles ever hit. These designers often tell *us* about things to watch out for.

We at R•E•P try hard to be fair and honest in our evaluations, pointing out the above-standard and the sub-standard performance attributes, keeping in mind the technical capabilities appropriate for a device's price range. We think we are fair, and in fact, probably bend over too far to be factual and objective: One recent review was pushed-back for nine consecutive months as we verified and re-verified the results with multiple qualified sources, including over a dozen calls to the factory. Yet, to some companies, merely pointing out blemishes, if they truly exist, is unacceptable. They are uncomfortable with anyone discussing the real, unpublished performance attributes of their product. Go figure.

So let me ask you: Does the pro-audio world need a magazine like R•E•P that seriously reviews audio products? That tells it like it is, technically speaking? That believes a working professional has the right to know the truth about the performance of equipment they might consider buying? We obviously think so. It's been a hot topic of conversation around R•E•P lately. You see, we like to discuss ethics and our responsibility to the industry.

But what do you think? Do you, the pro reader, want more tech and more truth, or more hype? Are you happy with only getting a marketing department's version of performance reality, the kind of thing another magazine might sneakily reprint as if they wrote it themselves? Or do you want to see more tech-intensive product reviews? Inquisitive minds want to know. ■

Mike Joseph
Editor

Letters

Hands On Response

From: Meyer Sound, Berkeley, CA.

We have read your review of our HD-1 Monitor ("Hands On," November 1991). There are significant discrepancies between your measurements and ours. We cannot address the reasons for this, since R•E•P did not describe or identify the testing facility, list the equipment configuration employed, name the consulting engineers involved, or make public any information that would permit an independent judgement regarding the resolution and accuracy of the measurements.

We will endeavor here to address some of the major points in your discussion of our product.

1.) Given appropriate phase correction, it is, indeed, physically possible for a 2-way loudspeaker to behave as a true point source within a specific coverage area. By definition, a point source radiates spherical sound waves uniformly such that, for each doubling of distance from the source, the sound pressure level drops by 6dB across the entire frequency band of operation. This is relatively simple to verify through measurement, as can be seen in Figure 2 of your review.

2.) Our frequency response specification is an accurate and honest representation of the HD-1's performance. Respected engineers in several different countries have succeeded in confirming it through careful application of appropriate acoustic instrumentation.

3.) The use of an anechoic chamber does not, in and of itself, assure adequate accuracy to produce valid measurements of the HD-1. Even perfectly anechoic rooms (<10% pressure reflection from walls, ceiling and floor) exhibit ±1dB residual amplitude error, and most anechoic facilities do not meet this standard. When attempting to characterize a speaker whose amplitude response falls within a ±1dB window, this magnitude of error is unacceptable.

4.) It is not physically possible for a loudspeaker of the HD-1's design to exhibit 340° coverage at 2kHz (as depicted in the directivity plot of Figure 6). Moreover, the polar plots of Figure 5 and 7 do not show anything close to this figure, at any frequency in either axis. These discrepancies should have been sufficient to cause the data in Figure 6 to be rejected.

5.) It is impossible to determine the validity and meaning of polar plots if the measurement distance and the decibel scale of the polar grid are not specified; the same is true of ETCs with no horizontal scale marks (Figure 9).

6.) The action of the HD-1's dynamic limiting circuitry, which functions on a cycle-by-cycle basis and only at the point of overload, cannot be accurately characterized with sine wave sweeps (Figure 10). Sine waves have the smallest crest factor of any test signal except square waves, and do not indicate, in any way, the dynamic response of a system to musical material.

7.) To reproduce the remarks of a competing manufacturer within the context of a product review is, we feel, inappropriate. The comments by Mr. D'Arcy of Miller & Kreisel regarding the effect of the HD-1's baffle on its tweeter are unscientific, and are not supported by measurements. Such effects have been known for many years, having first been described by Olsen. They are correctable in the transfer function of the system, and are dealt with electronically in the HD-1.

8.) We feel Mr. Levitin's comments that the HD-1 is not "musical sounding" or "fun to listen to" are so vague and subjective as to be meaningless. And just as "I Love Lucy" reruns look grainy on HDTV, poorly recorded material will sound bad on the HD-1. This hardly seems an indictment of our product, nor does it indicate that the speaker is suited for only one particular style of music.

Professional recording engineers and producers require highly accurate audio monitors if they are to produce product that will translate on a wide variety of consumer loudspeakers. That a majority of top-echelon audio professionals worldwide have selected the HD-1 as their primary (or sole) monitor is, we believe, evidence of this fact. The quality of their work speaks for itself.

Meyer Sound takes great pride in its products, and goes to significant lengths to provide complete and accurate data on the HD-1. All of our published data are fully documented, with test conditions clearly and completely specified. Those who wish to obtain this documentation are welcome to contact us directly.

Mike Joseph replies:

It is unfortunate that so much was misinterpreted by Meyer Sound concerning the HD-1 product review. Although we dedicated an unusually large amount of space to this high quality, most innovative and groundbreaking monitor speaker, due to space limitations we were unable to share *all* of the information we acquired (over 100 separate measurements, excluding the 38 graphs and charts contributed by the Meyer engineering department).

We felt we hit the major high points. In fact, many technical aspects were briefly touched upon which clearly could bear further investigation: a horizontal polar response which is wider at 5kHz than 1.5kHz; the large mid-range disparity in response linearity on-axis vs. 10° above axis; the strong down-firing lobe at the crossover frequency; the fact that Meyer claims linearity in their measurements only referenced to 0.5 meter, yet virtually all real-world nearfield listening is done at .75 to 1.5 meters distance (their own measurements show increasingly larger response aberrations beyond 0.5 meters); that at no point, even on their submitted measurements, is a true ±1dB response across the stipulated bandwidth indicated (we have never seen, anywhere, documentation showing same); that any device with a crossover and two separate, displaced, band-specific radiators cannot be a true single wide-band point-source across its entire spectrum, at all measuring distances; that the horizontal pattern generally (with some exceptions) collapses with rising frequency, which is normal, opposing their published claim of 60°, etc.

The point here is not that the HD-1 speaker is or isn't perfect, however you measure perfect. The point is that, at almost $5,000 a pair, they should be a known and definable quantity, with their limitations fully understood, expressed, and defined by the manufacturer via open discussion (short of publishing proprietary or patented design information, of course). Most companies provide this as a service to their customers.

More importantly, the speakers must be worth their value as a listening and mixing tool. Now, we *know* today's speakers have limitations — they're not electrostatically driven ionic plasma — they're electro-mechanical devices! The HD-1 is constructed, after all, of fairly traditional components — wood, paper, steel, copper, fabric, plastic — and as such, no matter the proprietary processing electronically added, suffers, as I said in the article, from traditional physical limitations. Pistons narrow their propagated pattern — it's a wavelength-related phenomenon. Drivers reach excursion limits. Signal compression and/or limiting affects response linearity. Two drivers with centers separated by eight or so inches do not a true point-source make across an entire frequency spectrum. Baffle edges contribute to diffraction effects. *All dynamic loudspeakers in boxes suffer these phenomenon.*

Specifically responding to the addressable points mentioned in the letter:

A) Information relating to the measurement devices and techniques applied are available in the November, 1990 issue of R•E•P, in the "Hands-On: Radian MS-8 Loudspeaker" review. As was pointed out in November, additional data was collected from other highly qualified sources using TEF, MLSSA, UREI, B&K and HP measuring systems in various and assorted environments, all compared by us for accuracy. Only material which correlated and was corroborable was used, although correlation between the various measurement techniques and locations was very high. We stand behind the measurement information presented. If anything, we can be accused of being conservative.

B) The definition of a point-source in the letter's item #1 is simplistic at best, and doesn't address wavelength, measurement distance, measurement angle, etc. The common acoustic definition of a point-source is a hypothetical point in space, *smaller than the wavelength of the sound being radiated*, which propagates omnidirectionally, in 3D. Few, if any, speakers are true point sources.

Any single component from a given multi-way speaker might be referred to as a point source if viewed at only one specific frequency and at a very specific coverage angle. But

Continued on page 8

Letters

Continued from page 6

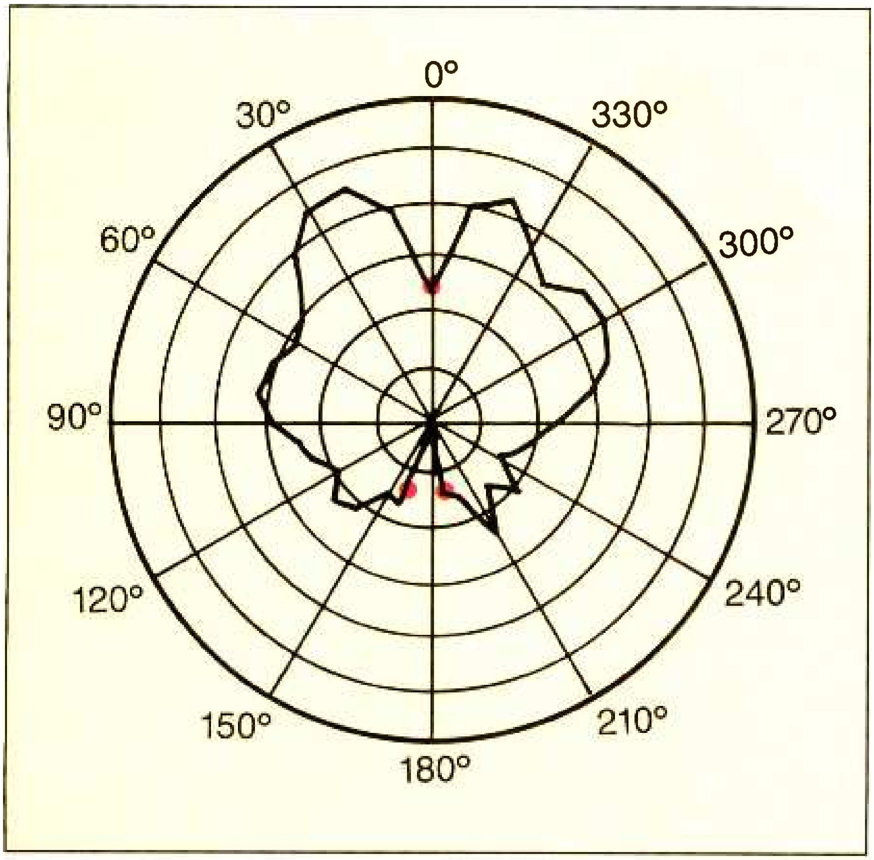
what does that mean? And does it fit any accepted definition? I'd like to hear from the readers on this!

C) As can be seen in the accompanying vertical polar plot, the measurements alluded to in point #4 were normalized on-axis at what happens to be the somewhat troublesome crossover area, in this case 2234Hz. Referenced to the cancellation-caused null, the –6dB down points are indeed 170° off-axis in both directions, for 340° coverage. Neither Figures 5 nor 7 show this worst case, but the 1.9kHz trace in Figure 7 of the review comes close. Also, the directivity plot mentioned (Figure 6) was made with 400 point measurement resolution for much greater detail, rather than the more typical 1/3-octave band plots. (Note: the pattern shown is a good example of the HD-1 *not* acting like a point source.)

D) As is customary, the polar grid decibel scales are 6dB/div. in Figures number 5 and 7 in the article. Measurements were made at 3 meters and normalized to 1 meter when appropriate. The Figure 9 scale is 1,000 Sec, as indicated. Our honest apologies for not captioning the original graphs in a more detailed manner. We promise that this will be corrected in future reviews.

E) In reference to point #6 in the letter, we should mention that we highly qualified the validity of sine sweeps on dynamic limiting in the original article. We suggest you please re-read the paragraphs on page 62 beginning, "It is important to emphasize, despite the radical appearance of the traces in Figure 10, that swept sine waves are not music or voice ......"

In conclusion, we feel, as do the many who have responded to us after the publication of the review, that the Meyer HD-1 is an excellent monitor speaker with specific unique characteristics. The measurements speak for themselves. The HD-1s readily lend themselves to a range of high-quality audio applications. Obviously, they aren't for everyone, or for every purpose. Few things are. We highly recommend that every reader listen to the speakers personally and draw their own conclusions.



Vertical polar plot of Meyer HD-1 showing – 6dB down points at approximately 170° off-axis, up and down. Note 0° reference point on null. Measurement taken on cabinet axis, 3 meters, at 2234Hz, in a full anechoic environment using a B&K 4133 mic fed into a Techron TEF system, measured at 10° increments. Additional equipment included a B&K 2610 measuring amp, a UREI Model 200 chart recorder and a Hewlett-Packard 3325A function generator. Decibel plot scale is 6dB/division.

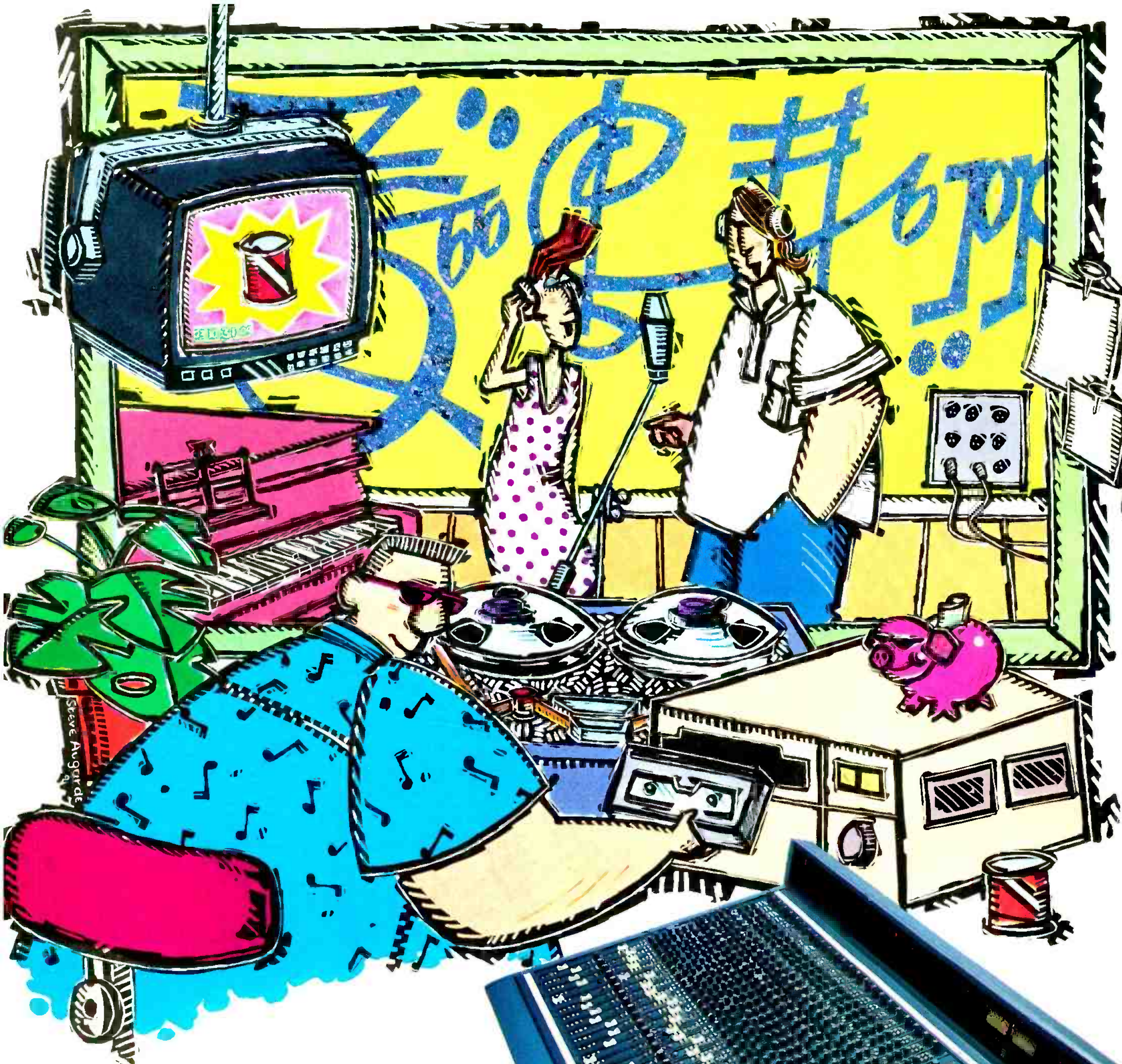
Dan Levitin Replies:

I feel that equipment reviews all too often ignore that audio equipment needs to be used by people with ears. R•E•P incorporated my subjective feelings about the HD-1s in a sidebar, separate from the technical portion of the review, and this seems like an appropriate treatment. You state that my comments are "subjective" as though this is an indictment. The whole point of my piece was to offer a subjective view of what it is like to actually use and listen to the speakers, a point which I'm sorry you missed.

I interviewed two people whose musical judgement I hold in the highest esteem. Jeffrey Norman is a top recording engineer, and as such, I felt his opinions about what it's like to actually *listen* to the speakers would be of interest to R•E•P readers. John D'Arcy is not a Meyer competitor; he is currently an electrical engineer in an unrelated field. Previously, he was a key player in the design of one of the most influential loudspeakers of the last 20 years. The opinions of these two professional listeners were indeed "unscientific" and that is precisely why I wanted them. Listening is what recording is all about, and these are two of the best listeners I know.

No one mixes in a vacuum. Engineers and producers routinely listen to other projects as a reference while they are mixing. Many people have played recordings that sound terrific on all other monitors, but sound strange on HD-1s. This can be disorienting and scary. Readers should know that other people have had this experience. Nowhere in the sidebar, however, did anyone say the Meyers weren't appropriate for mixing, or that they weren't accurate, only that they weren't ideal for *listening* in some people's opinion. I know of no professional engineer — or musician, for that matter — who lives by specifications alone: the *feel* and *sound* of equipment are the ultimate criterion. ■

Send letters to R•E•P, Box 12901, Overland Park, KS 66282; or fax 913-541-6697. Letters must be signed and may be edited for length and clarity.



Perceived Value

Some of the below from Peter Watson, *New York Times*.

Item: Eddie Murphy forks out $26K+ for vest and black suede headband worn once on stage by Jimi Hendrix.

Item: Collector pays $14,780 for handmade Christmas card from John Lennon to his first wife, Cynthia.

Item: Madonna's soiled baseball uniform from unreleased film "A League of Their Own" nets $2,100 at auction.

Item: "Reserved" table card with message "For guests of the Bruce Springsteen band" is purchased for $990.

Message: Check your dumpster and keep that dirty hanky from the last session!

Roll Your Own Label?

Vanity recording was recently summarized by the *Wall Street Journal* this way: "The economics are steadily improving. Today, musicians say, it requires only a few thousand dollars of digital equipment to capture sound quality that could once be achieved only at major studios. In the last five years, the production costs of a typical quality CD has dropped as much as 50%. At the same time, cheap desktop computers are assisting musicians to write music, design album covers, publish fan newsletters, track cash flow, and maintain mailing lists, among other things."

Instant Record Label?

As evidence of this trend, the *WSJ* cites bands that include former Electra records trio Shoes, which last year approached a $250K gross on its own Black Vinyl Records. On the other, probably more common end, the *Journal* referenced industrial artist Claude Wiley, who had to delay upcoming projects while waiting on a $1,500 outstanding debt from a record distributor. Clearly a cash flow susceptible proposition.

"Desktop recording labels" might seem in opposition to the mega-corporate recording industry. Certainly, the big studio project studio debate is tied to this trend. But positive long term effects are likely for all parties. A good way to think about desktop record production is that "homegrown labels" will assume a minor league posture within the music business. The quantity of non-commercial or fringe music gaining market access (however small) is absolutely beneficial. And the same engineers who mix and master successful desktop projects may soon be bringing honed chops to big rooms. ■

PEOPLE

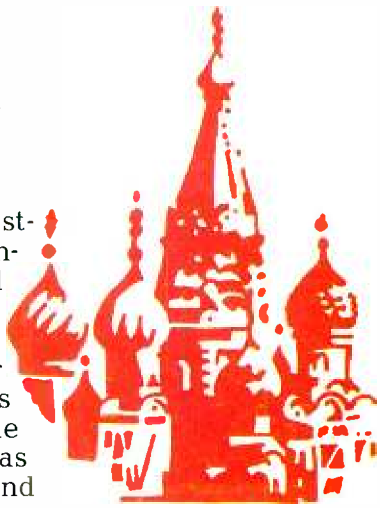
In order to pursue their music industry careers unemcumbered by studio ownership, **Britt Bacon** and **John Eden** recently closed their Topanga Skyline Recording Company ... **Carl Reavey** has assumed the position of general manager for all Amek and TAC western hemisphere operations, while **Lewis Frisch** was been appointed regional Amek and TAC sales manager for U.S. operations ... Otari Corporation has promoted **Emil Handke** to national sales manager. **Lee Pomerantz** is Otari's new export sales manager. **Roberta McKeehan** has been named industrial product sales coordinator at Otari ... **Karen Spriggs** has joined Sony Broadcast and Communications as public relations officer ... **Dr. Mike Smyth** has been named to head the startup of Audio Processing Technology's Los Angeles sales office ... Neotek has announced that **Julie Bacher** has been promoted to business manager and **Tom Lay** has been named sales and marketing group leader ... **Dave Collie** has been assigned new manager of SSL's western operations ... Electro Voice has appointed **Neil Andison** general manager of Mark VI Audio Canada ... **Hudson Fair** of Ealing Mobile Sound was elected chairperson of EARS (Engineering and Recording Society) in Chicago for 1992. ■

Quote Book

"The generalist knows less and less about more and more so that eventually he knows nothing about everything. The specialist knows more and more about less and less so that eventually he knows everything about nothing."

— Fred Harbert, uncle of unnamed East Coast record producer.

trend watch



JUST WHAT THEY NEED: The post-Marxist avalanche of decadence continues as a new "independent sound equipment and musical monthly" named *IN/OUT* is scheduled for publication next month in Russia. Editor Michael Subbotckin, said, "*IN/OUT* is dedicated to reporting and analyzing the sound equipment industry and market as they face a new era full of uncertainty and opportunity."

VAPORWARE: Tandy Corporation's long awaited compact disc player/recorder seems to have a blue smoke element. Since early 1988, Tandy has struggled to deliver its THOR (Tandy High-intensity Optical Recorder) which promised CD read/write capabilities for less than $500.

It didn't happen for at least two reasons: First, Tandy insists the ongoing copy protection battle intervened. Secondly, and realistically a bigger obstacle, THOR could write-erase-read only 10,000 times before disc crash. That limitation effectively removed THOR from any computer storage applications.

Meanwhile, Sony has announced development of a $500 magneto-optical technology which will tolerate millions of write-erase-read sequences.

INVITATION TO A SIDESHOW: As Dire Straits began its fall (1991) tour, concert site demonstrations were planned for Philips' new digital compact cassettes (DCC). By sponsoring the 300-city tour, Philips apparently hopes to jumpstart the new format for the inevitable marketplace confrontation with Sony's DAT technology. Let the battle begin.

ANOTHER ONE BITES THE DUST: It is with sadness that we note the passing of Freddie Mercury. Born in Zanzibar as Frederick Bulsara, Mercury died Nov. 24, 1991 in London from AIDS complications. Flamboyant and controversial, the long-time Queen lead singer maximized his 45 years with unrivaled bravado.

DID YOU KNOW: The United States Copyright Law (Title USC 101) identifies a "public performance" as one that occurs "at a place open to the public or at any place where a substantial number of persons outside of a normal circle of family and its social acquaintances is gathered." As such, the act of broadcasting, cablecasting, communicating a performance to the public , or even playing music-on-hold on a phone system is considered "performance" under copyright law. ■

Random Access

STUDIO UPDATE

Facility/Location	Details
NORTHEAST	
Hit Factory/New York	Studio A2 received retrofit of SSL's Ultimation console automation system to its 64-input SL 4000 G series desk.
Acme Recording Studios/ Mamaroneck, NY	Has upgraded its facilities with a second Otari MTR-90, MKII 24-track machine and Lexicon 480L digital effects processor.
Golden Studios/ Hancock, NH	New studio, designed by chief engineer David Torrey, features the Spectral Synthesis Digital Studio.
SOUTHEAST	
Ultrasonic Studios/New Orleans	Expanded to 48 tracks with its recent addition of a new Studer A827 24-track recorder with 24 channels of Dolby SR.
The Music Mill/Nashville	Installed the Focusrite recording console with GML automation into Studio A.
New River Studios/Fort Lauderdale	Recently retrofitted flying fader automation to their Neve 8108 console and added a Mitsubishi X850 32-track digital tape machine with Apogee filters.
Omega Studios/ Rockville, MD	A Mitsubishi X-850 32-track digital recorder has been added to the Studio A SSL system.
Windmark Recording Studios/ Virginia Beach	New digital mastering facilities anchored by a Sony SDP 1000 and a Studer Dyaxis.
MIDWEST	
Triad Productions Des Moines, IA	Completed main room installation of the new Euphonix CSII digitally controlled analog studio system.
Zeta Recording Studios/Toledo, OH	Installed a 32-input Hill Concept 8400 console.
SOUTHERN CALIFORNIA	
University of California at San Diego/Burbank	Recently purchased an API 48x24 Discrete series "Touch Reset" console.
Kingsound Studios/North Hollywood	Installed a Neve V3-48 console with 60 channels of flying fader automation, 12 custom inputs with choice of API 550A, 550B and Massenburg 8200 EQ. A Studer A827 24-track and an Ampex ATR 102 2-track were also added.
NORTHERN CALIFORNIA	
Music Annex Post Production/ San Francisco	Has added a second New England Digital Post Pro editing system.
UNITED KINGDOM	
BBC Radio/London	Has ordered an AMS VCS, 48-fader assignable console for the Maida Vale studios.
CTS Studios/London	Has invested in a 60-channel Neve VRP console with recall and flying fader.
DESIGNERS	
Walters-Storyk Design Group/ New York	Currently working on two new conference/listening rooms and a suite of offices for Mercury Records Worldwide Plaza location in Manhattan.
Harris Grant Associates/United Kingdom	Has been appointed by Sony Classical Productions to construct two new mixing/editing suites in its Manhattan engineering facility.
Russ Berger/Dallas	Completed design of a personal recording studio addition for the residence of producer/engineer Mike Galesi in Bernardsville, NJ.

NEWS NOTES

President George Bush toured **Peavey Electronics** and addressed nearly 2,000 Peavey employees on a December 3, 1991 visit to Meridian, MS. While acknowledging the prolonged downturn in global economy, President Bush pointed to Peavey as a prime example of American industry's potential for international growth through an aggressive export posture.

As a memento of the trip, Hartley and Melia Peavey gave the president one of their new line of electro-acoustic guitars. The customized red, white, and blue axe was appropriately named "The Chief." [See Feb. 1992 Random Access for further coverage of the Bush visit.]

The Society of Professional Audio Recording Services (SPARS) board of directors decided to retire the SPARS code at a board meeting held during the 1991 AES convention in New York. The SPARS code, introduced in the mid-'80s, was used to identify which portions of the recording process were digital and which were analog.

Ampex Corporation and Sprague Magnetics have reached an agreement that will allow Sprague to provide ongoing worldwide service and support for Ampex audio recorders.

Robert G. Shaw CEO of **International Jensen Incorporated**, recently announced the formation of the corporate technology division, a group dedicated to advanced research in audio and sound reproduction.

TimeLine and Digidesign have formed an alliance based on TimeLine's new video clock card, known as the Pro Tools Interface. The card allows Digidesign's Mac-based production system full multimachine synchronization.

Focusrite Audio Engineering, Bourne, England, has signed a distribution pact with George Massenburg Labs of Los Angeles, for GML to serve as sole North American distributor of Focusrite studio consoles.

API Audio Products and SONTEC have entered into a licensing agreement to begin marketing the new API 554B parametric equalizer.

Rivera Research and Development offically joins the JBL Professional family during the winter 1992 NAMM conference.

ADDRESS CHANGES

Beyerdynamic has moved to 56 Central Ave., Farmingdale, NY 11735; 516-293-3200; fax 516 293-3288.

The new corporate headquarters for **Washburn International** is 255 Corporate Woods Parkway, Vernon Hills, IL 60061; 708-913-5511; fax 708-913-7772. ■

Fresh Tracks

By Dan Levitin

The Benny Green Trio: "Greens"
Label: Blue Note
Produced by: Matt Pierson
Engineered by: Jim Anderson
Recorded at: Power Station (New York)
Mastered by: Ron McMaster at Capitol
SPARS Code: AD

Comments: Green is one of the most exciting new piano players we've heard in years. This album of straight-ahead jazz is a model of jazz trio recording at its best.

Of special interest: The instruments are beautfully recorded — very clear and bright. The ambiences are very natural. All of the instruments are given their own position in the soundscape. The piano is remarkably even throughout its tonal range, and the drums present and well-blended. Recorded direct to 2-track, 15ips, 1/4-inch analog using Dolby SR. To these ears, the fidelity is indistinguishable from the best digital. Definitely worth a listen.■

Dan Levitin is R•E•P's music production editor and a producer based in Stanford, CA.

Nancy Argenta with the Chandos Baroque Players: "Alessandro Scarlatti Cantatas"
Label: EMI Classics
Produced by: Nicholas Parker
Engineered by: Tim Handley
Editor: Adrian Hunter
Recorded at: Concert Hall (University of Cardiff)
SPARS Code: DDD

Comments: Argenta possesses a remarkable voice, not just for her technical facility, but for her ability to blend so well with the chamber orchestra. The timbre of her voice weaves in and out of the instruments in just the way the composer must have intended. Sonically, the most amazing thing about this recording are the ambiences from the conert hall. Warm and full, it sounds as if only ambient mics were used, without even a spot mic on the vocals (or one which was barely used). The recording realistically conveys the sense of being in the audience during the performance. ■

REISSUES AND COMPILATIONS

Stan Getz: "The Best of the Verve Years, Vol. 1." We've always felt Getz's best recordings were done during this period and they are assembled here, including his sessions with Bill Evans, Jobim, Dizzy, Chick Corea and Charlie Byrd. Getz has never sounded better than on the tracks compiled and collected herein. The tapes sound as if they were recorded just last week. Engineer Phil Schaap works restoration magic on tracks dating back to 1952.

• • •

Les Paul: "The Legend and The Legacy." (Capitol) If there is one single individual all of us owe our careers to, it is of course, Les Paul, inventor of multi-track recording, flanging, variable recording speed effects and a host of other studio mainstay techniques. This beautiful 4-CD box is a must-have for everyone in the recording business, not just for the historical value and inspiration it provides, but for the four hours of great and innovative music. Produced by Ron Furmanek, remastered by Bob Norberg and Rus Paul.

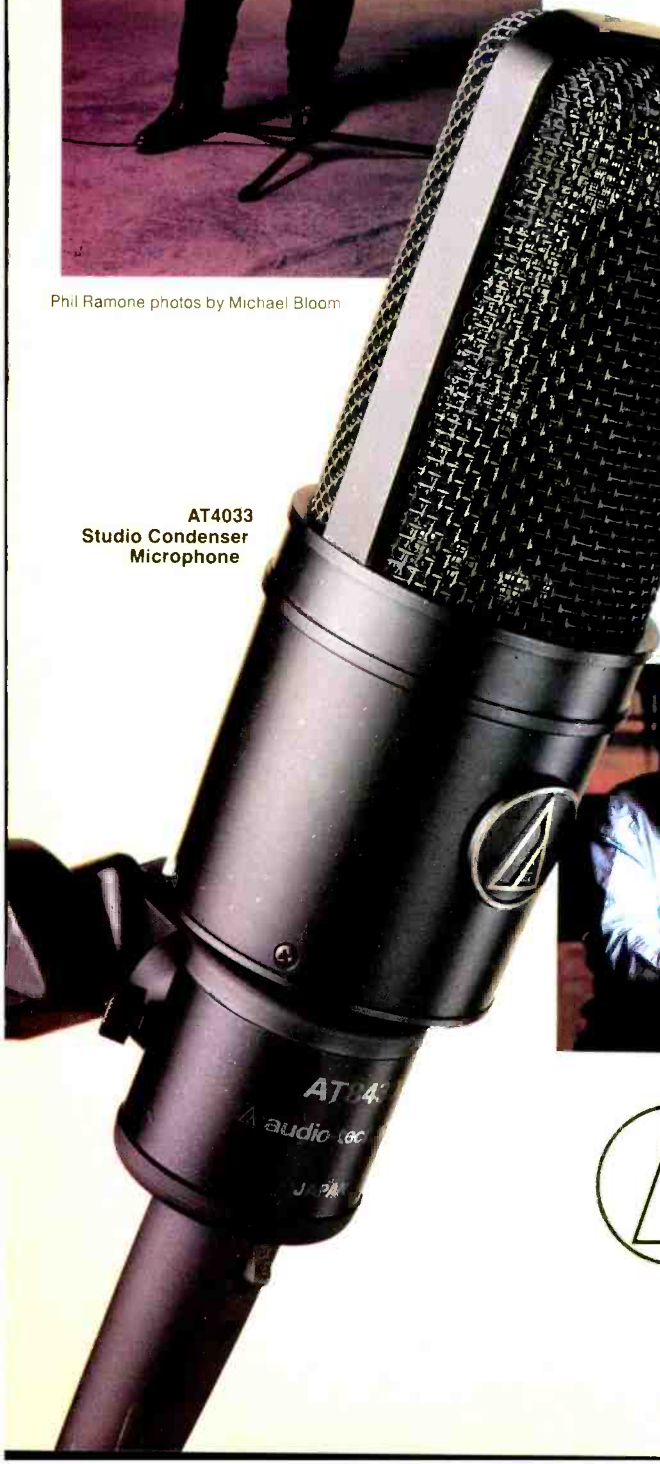
• • •

Howlin' Wolf: "The Chess Box." Another superb boxed set from MCA reissue producer/wizard Andy McKaie, this 3-CD set captures the best of one of the great blues artists of all time. The packaging and sonics are first rate, and the liner notes informative. ■

Phil Ramone photos by Michael Bloom

We asked Phil Ramone to field test our new AT4033 studio condenser microphone.

He wouldn't give it back!



AT4033
Studio Condenser
Microphone

Phil Ramone knows exactly what he wants from a studio microphone. And when he tested a sample of our new AT4033 cardioid condenser microphone, he knew it was right for him and ideal for the artists he records.

He liked being able to concentrate on getting the right music from the musicians, rather than first spending time experimenting with EQ to get the right sound.

What Phil Ramone heard was the result of new condenser technology inside the AT4033. The diaphragm is only 2 microns thick, with a vapor-deposited gold conductive coating just 50 angstroms thick. This sophisticated, very low-mass diaphragm is aged in a five-step process that insures unchanging performance for years.

The high head-room and wide dynamic range, plus low noise floor, make the AT4033 ideal for the most demanding digital recordings. And the maximum input SPL is an awesome 140 dB, so important when recording high-output instruments and very close-up vocals. In addition, transformerless design contributes to overall sonic transparency. The AT4033 also includes a switchable 10 dB pad and lo-cut filter, plus a built-in pop filter and internal shock mounting.

We're not certain we'll ever get the sample AT4033 back from Phil Ramone, but no matter. We're busy making your AT4033 right now. For more details on this impressive new microphone, ask your A-T sound specialist to schedule a test of the AT4033 today.

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Circle (9) on Rapid Facts Card

Fresh Tracks

The Modern Mandolin Quartet: Tchaikovsky, "The Nutcracker Suite"
Label: Windham Hill
Produced by: Mike Marshall
Engineered by: Oliver DiCicco
Recorded at: Mobius Music (San Francisco)
Mastered by: Bernie Grundman
SPARS Code: AD

Comments: The ambitiousness of transcribing "The Nutcracker Suite" for four mandolins is mind-boggling and the result is truly brilliant. The transcription (by MMQ members Paul Binkley and John Imholz) not only works, it excels, adding unanticipated newness to the music.

Of special interest: Oliver DiCicco's recording is flawless. His use of stereo imaging and his sense of tone at once blend the instruments just the right amount, while allowing them to maintain the essence of their distinctiveness. And DiCicco is flexible — his past projects include engineering for The Dead Kennedys as well as Michael Hedges' "Aerial Boundaries" (for which he received a Grammy nomination for best engineered recording). ■

FOCUS:

OLIVER DICICCO, Engineer, "The Nutcracker Suite"

R•E•P: How did you mic the musicians?
OD: Each instrument was recorded with a stereo mic pair, so there were eight mics close-miking the band, and a pair of B&K 4006s that were used for ambient mics. Mike Marshall's mandolin was miked with a pair of Neumann KM54 tube mics; the second mandolin and mandola with KM84s modified by Klaus Heine. Mike has a couple of Klaus Heine KM84s too, and the mandocello was recorded with one of his KM84s on the neck and a tube U47 on the body. We recorded direct to stereo at 15ips using Dolby SR on a $^1/_4$-inch Studer 820, with Agfa 468.

The record was done without any EQ at all, and we went for tone by placement of the mics. We went through the mic pre's on my Neve 8068, and we went for the minimal signal path that was available with this particular console and configuration. I put the mics in approximate positions, and then Mike went out into the studio with headphones and he'd move the mics to the point where he wanted them to be. He likes a certain brightness and a certain presence to the instruments, but he wants them to be blended as well.

R•E•P: How did you keep such good separation between the instruments?
OD: The musicians stood in a semi-circle, roughly 15 feet in diameter. They were in fairly close, to allow them to interact with each other the way they normally do on stage.

R•E•P: What about moving around? There's never a time on the recording where it sounds like their tonality changed because of moving away from or toward the mics.
OD: They're pretty disciplined as a group and they all have recording experience — they're aware of mic placement.

R•E•P: What did you use for reverb?
OD: We used an early Lexicon 224 with some of the original programs, and an Ultra-Harmonizer H3000SE on a modification of the concert hall program. We changed the parameters to match the physical dimensions of the room in terms of pre-delay, to get the reverb to sound fairly natural. The other thing is, we used the room mics as the primary reverb sends; the idea was to try to create as natural a sound as possible but still maintain the presence that Mike wanted, which is kind of mutually exclusive.

R•E•P: With all those mics in such close proximity, weren't you worried about phase problems?
OD: You do have phase information going on all the time. The way I look at phase is that if it sounds good and it still sounds good in mono, then the phase information is part of the sound. If you have more than one mic on an instrument you're gong to have phase information present, but that's what gives the instruments a sense of space, as opposed to having a single point of sound emanating from a speaker. You create a dimension for that particular instrument — you give it some size.

Basically, we're dealing with an illusion, but it's trying to create a good illusion of space while still maintaining a sense of definition and presence. The [phase] leakage works to your advantage — it adds more dimension to the sound. ■

Fresh Tracks

Keith Whitley:
"Kentucky Bluebird"
Label: RCA
Production director: Garth Fundis
Produced by: Garth Fundis, Blake Mevis, Keith Whitley, Kix Brooks, Fred Koller, Don Cook
Engineered by: Bill Harris, Gary Laney, Garth Fundis
Mixed by: Garth Fundis
Recorded at: Music City Music Hall, Woodland Sound Studio, Sound Emporium (Nashville)
Mastered by: Denny Purcell at Georgetown Masters
SPARS Code: N/A

Comments: When Whitley died a couple of years ago, he left behind tracks "in the can" in various stages of completion. Hardly a "dregs" album of previously forgotten tracks, "Kentucky Bluebird" may contain Whitley's finest material ever. Fundis tastefully rerecorded some of the backing tracks to accompany Whitley's vocals. The result is a cohesive and poignant retrospective of this highly talented country vocalist.

With the recent death of such greats as Whitley, the Vaughan Brothers and Miles Davis, it gives one cause to stop and reflect about the importance of the recording engineer's job, capturing a moment in musical time to allow it to live on for posterity, and in some cases, documenting the great musical artists of our day.■

FOCUS:

GARTH FUNDIS, Production Director, "Kentucky Bluebird"

R•E•P: How much of the original tracks for these songs did you keep and how much did you scrap?
GF: "Going Home" for example was a complete restoration, where I only kept the vocals. I took the 24-track analog masters and transferred them to 32-track Mitsubishi 850, and then basically we put down the new click track, with Eddie [Bayers, drummer] playing hi-hat to the old track. That was the first pass. Then for the second pass, everyone played at once. I didn't do this piece-by-piece — I tried to do this the way we would have done it normally, when Keith was there. That is, to have everyone there playing on the session live, all the musicians in the studio at the same time. When solos come up, 'go for it', you know, don't lay out and overdub it later. In some cases we did go back and work on solos, but we were always trying to go for it.

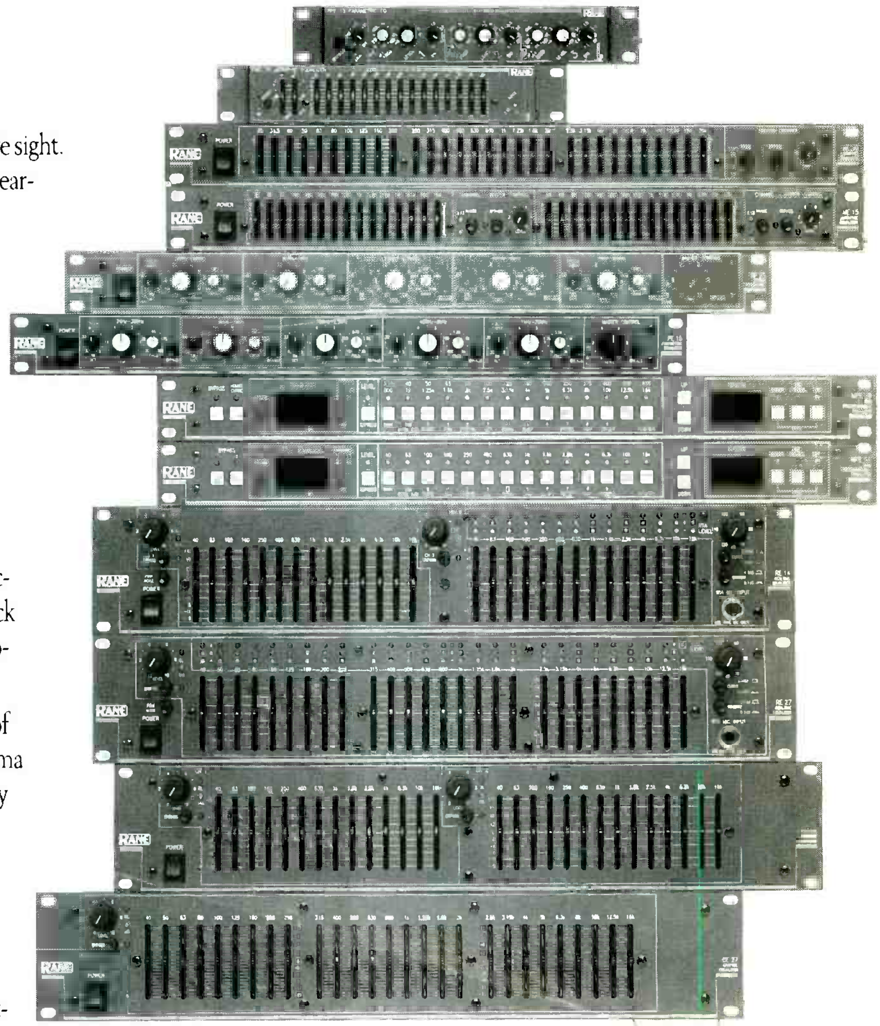
The other tracks that were complete restorations were "Somebody's Doin' Me Right," "Brotherly Love," "Kentucky Bluebird" and "I Want My Rib Back." The latter two were originally demos from Tree Publishing that Keith had sung early in his career. The other songs had originally been produced by Blake Mevis, who had produced two of Keith's earlier albums, but those songs had gotten shelved because Keith and the label weren't happy with them the way they were.

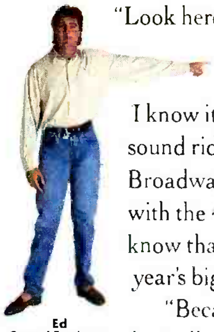
R•E•P: You take production and co-engineering on most of your projects. How do you decide when to engineer and when not to?
GF: During live sessions there are so many things to take care of as far as communication with the artist and the players, that I really don't feel like it's fair for me to try to do all of it and make everyone wait while I'm working on a drum sound or trying to change a mic or something. I work closely with my engineer Gary Laney, but I don't actually sit behind the console during tracking. But I do all my own mixing.

R•E•P: What reverbs did you use on "Kentucky Bluebird"?
GF: For vocals the EMT 250, I guess set at 2.2 sec. Sometimes I put a tight pre-delay on the voice, of about 60ms I use the AMS reverb on the ambience program for some of the drums, a couple of PCM70s, Rev 7. I used an old tube Fairchild compressor on the vocals — I really love the sound of those things, they just give it a 'sound.'

R•E•P: You mixed on a Neve?
GF: Yeah, I do all my mixing on the Neve 8128 in Studio A at the Sound Emporium. For years they had a Harrison console which I refused to use — it was awful. I hate VCAs, which is why I haven't used SSLs much and when I do, I try to bypass the VCAs. People have tried to persuade me over the years that the new VCAs are improved, (that) you can't tell they're VCAs anymore. That's fine for them, but I prefer the Neve sound. I've used Sontec mic-pre's from time to time, but I have no complaints at all about the Neve pre-amps.

[Fundis has also produced recent albums by Trisha Yearwood and Don Williams.] ■





Ed
Sound Engineer

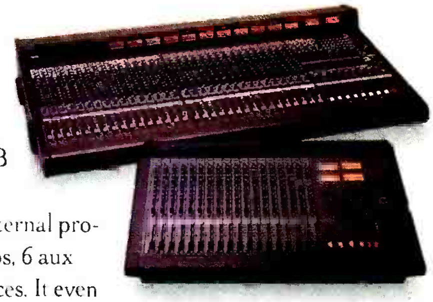
"Look here, I know the PM3000.

"I know it's at the top of the list of the best live sound-reinforcement consoles. I know it's written into all those big concert tour sound riders. I know it's in the major theaters on Broadway. I know it's in the 5,000-seat churches with the 400-seat choirs. And I also happen to know that it's in all those T.V. trucks producing this year's biggest sporting events. And I know why.

"Because the PM3000 is flexible. Because it's logically put together. Because it performs. Because it's a pleasure to use. Because everyone likes working with it.

"But, here's the news.

"There are two more PM series consoles. And they start at a mere $5,500 MSRP. So obviously, they're for those situations where you want the best console available. But you don't have the space or the budget to get the 3000.



"The PM1800A was just updated. So it has an improved signal-to-noise ratio (6 dB better). And 0dB insert points for easy gain matching with external processors. It's got 8 groups, 6 aux sends and 4 mix matrices. It even has the same mute grouping feature you find on the 3000. But that's not the end of it.

"The PM1200 has the same roots. But in a more compact format. It's got 4 groups plus stereo, 4 aux buses, and 4 mute groups. You can get 16, 24, or 32 input channels and you still get two additional full-function stereo input channels.

"Obviously, they're both ripoffs of the Yamaha PM3000."

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"Obviously, they're both ripoffs of the Yamaha PM3000."

Ed and his opinions are a composite of several obviously brilliant and successful sound engineers. Professional Audio Products, P.O. Box 6600, Buena Park, California 90622-6600. (714) 522-9011.

Circle (11) on Rapid Facts Card

Be There Now

By Tom Scott and Tom Kobayashi

There's a technological revolution taking place that has been creeping up so slowly, you might not have noticed it. But, in the remaining years of this millenium you won't be able to avoid it if you're in the entertainment business. What we're talking about is the inexorable movement to connect all of our facilities together into a worldwide shared workspace, that will eventually transform the business of recording and post-production.

External conditions are conspiring against the status quo. As you're surely aware, schedules and budgets are getting tighter and tighter. Any way to cut expenses is fair game. Travel is increasingly expensive, across town or across country. Cities have become gridlocked to the point where it may take two hours to deliver a tape across town. Between cities there are fewer flights and they're more crowded. The price paid for a cross country flight also includes an unhealthy dose of stress and airline food. As city life becomes more difficult, the high-priced performing talent prefers to live farther away, sometimes way out in the country. Producers have to pay more to bring them in for a session. All of these conditions argue for a better workstyle, one that progresses beyond the traditional studio work scheme — the one where everyone important to the process must be scheduled together in one room to do the work, whether it's for a meeting, a mix, an overdub, a voiceover, a mastering session or a playback.

While we've been grumbling over the price of gas and the traffic on the freeway, technology has been getting ready for easier collaboration over long distances. Remember when, not too long ago, if you wanted to send a schematic, a score, or a budget to a colleague, you photocopied it and put it in an envelope with a 13-cent stamp? In a couple of days it was probably delivered. Last year you faxed it for telephone comments. This year you are just as likely to E-mail a file and have the corrected score modemed back. Next year it's probable that your PC will call your colleague's Mac and you'll both work on the document simultaneously. Remote collaboration is the key concept here, and there are signs everywhere that it is a revolution whose time has arrived.

Here's an interesting experiment you might not have heard about: In Northern California, our local SMPTE chapter has members spread out over a 500-mile area from Eureka in the north and Fresno in the south to Reno in the east. That's too far to expect more than local folks to attend monthly meetings. So, throughout 1991, the enterprising program committee used Hi-8 video cameras and a news-gathering truck to broadcast the 2-hour meetings via satellite. Members in Sacramento or Ukiah get together at the homes of local members who own an ordinary satellite dish, participate in the meeting over beer and pizza, and call in questions on an 800 number to the meeting chair. Of course, you could tape the meetings and send out VHS copies, but the simultaneous remote collaboration is the key to a real meeting. Otherwise it's just a taped lecture.

Another related tip-of-the-iceberg development: Announcers and other voice talent have begun using satellite links and telephone program lines to record their lines for advertising agencies and television stations. For example, rather than paying to fly a famous announcer from Los Angeles to New York for a few hours work on a car comercial that is due in 24 hours, a Manhattan ad agency books time at a nearby studio, which in turn schedules satellite time and books a second studio in L.A. The talent arrives at the L.A. studio while the producer or director goes to the N.Y. studio. The latest script is faxed to L.A. minutes before the session begins. The studios patch from L.A. to N.Y. with a high-quality, one-way satellite audio link for the recording, and the director makes comments via an ordinary long distance call over a speaker-phone. An hour later, the ad is complete and the talent is off to another session. The producer pays for two studios and some satellite time, but is overjoyed to have beaten the client's deadline and incurred no travel expenses or per diem.

It's a simple step from that voiceover scenario to music overdubs. Imagine that you're a record producer (or maybe you are one). For about the cost of a one-way flight to Muscle Shoals, you can stay in your own town and overdub those horns remotely. You save on hotel, rental car, per diem and stress. You won't miss that emergency meeting for your next job, or worse, have to tell your kids for the second year in a row that you can't make the Cub Scout banquet or their birthday party. The musicians will have similar savings in time, stress, and family angst. The studios are happy with the collaboration because it means more work, and in some cases, work that would never have occurred without remote cooperation.

Long distance collaboration is already a proven option. At Skywalker Sound, we have been doing remote mixing, playback and ADR sessions via digital telephone tielines connecting our Northern and Southern California facilities. For more details, refer to Rick Schwartz's Digital Domain article, "The Audio WAN" (R•E•P March 1991). In that article, Schwartz may have coined a new buzz-word for the '90s. In computer parlance, a WAN is a Wide Area Network. What seems to be building here is a movement toward the Worldwide Entertainment Business Wide Area Network. Say it all together now: "The WE-B-WAN." That's it! We Be One! Wow, cosmic!

Seriously, though, there are powerful foces at work here. The telecommunications and computer giants are working toward providing the interconnection tools for us, but we must be smart enough to use them to the best advantage. The price of high bandwidth communications is coming down. Low bit-rate coding methodologies are evolving that will deliver the high quality demanded by the recording industry over inexpensive telephone lines. The proliferation of optical fiber networks will provide terrestrial digital paths that can carry not just audio, but video teleconferencing that the average studio owner can finally afford.

The economy and the march of urbanization have us squeezed between the hard place of budgetary cutbacks and the rock of traffic gridlock. We want to be able to have more family time and less travel time. Can this be the advent of the entertainment business version of telecommuting? Perhaps, instead of building another 24-track mix room, your next studio expansion should be a transcontinental overdub room or a teleconferencing setup.

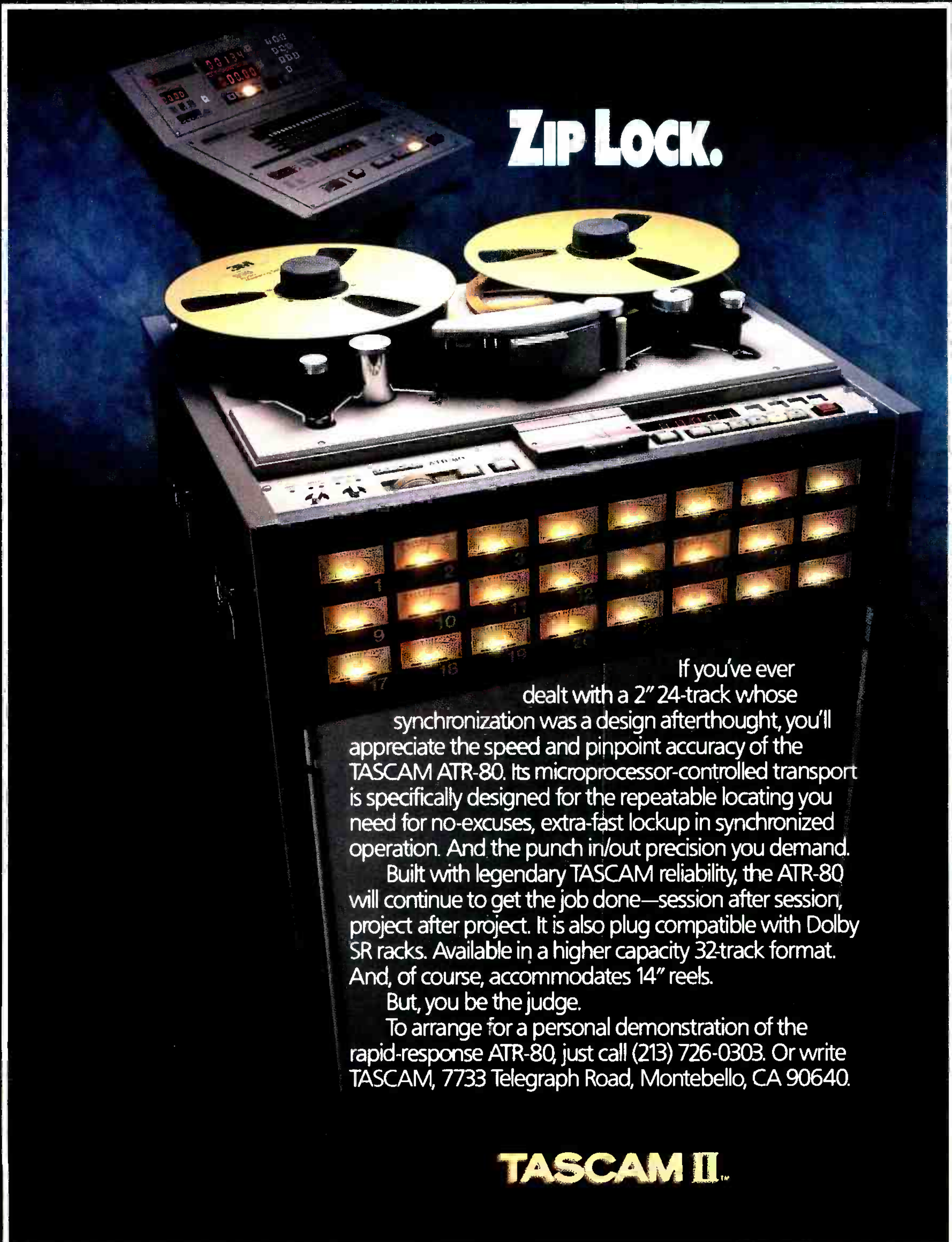
Beyond the personal life and company business benefits, there are world economic implications here as well. Communicating in place of commuting will save resources and cause less pollution. There might even be tax savings in some situations. If you take a tape overseas and overdub on it, you may have to pay a Value Added Tax. If you record remotely by satellite or telephone, the data contained in the call is not taxed (yet).

The most successful companies today are those that are thinking years ahead and forming alliances. They see the futility of short term defensive isolation and the benefits of a long term, global strategy. The current world economy rewards companies that have stretched out from their local bases to participate internationally. The American Entertainment Business is one of our most successful and far-reaching international exports. The power of these interconnections may help us maintain that preeminence while allowing us to participate in global entertainment markets that are far beyond our present reach.

Reflecting the SPARS goal of industry networking, we've tried in this article to point out a few of the early harbingers that point the way to our future as surely as the first few swallows into Capistrano mean "they're ba-ack." The coming years will see these small steps dwarfed by a giant networking growth allowing collaborations that would have been unthinkable a decade ago. We will enjoy improvements in the quality of our lives, while saving resources, time and money. The wise studio owner, engineer, and performer will investigate these new techniques and start thinking of ways to put the technology to work. ■

Tom Scott is director of engineering and Tom Kobayashi is vice president/general manager at Skywalker Sound, CA.

The Society of Professional Audio Recording Services is the industry's best source of business information. For details on membership or activities, contact SPARS at 4300 10th Ave. N., Suite 2, Lake Worth, FL 33461; 407-641-6648; fax 407-642-8263.


ZIP LOCK.
If you've ever dealt with a 2" 24-track whose synchronization was a design afterthought, you'll appreciate the speed and pinpoint accuracy of the TASCAM ATR-80. Its microprocessor-controlled transport is specifically designed for the repeatable locating you need for no-excuses, extra-fast lockup in synchronized operation. And the punch in/out precision you demand.
Built with legendary TASCAM reliability, the ATR-80 will continue to get the job done—session after session, project after project. It is also plug compatible with Dolby SR racks. Available in a higher capacity 32-track format. And, of course, accommodates 14" reels.
But, you be the judge.
To arrange for a personal demonstration of the rapid-response ATR-80, just call (213) 726-0303. Or write TASCAM, 7733 Telegraph Road, Montebello, CA 90640.
TASCAM

By Rick Schwartz

Digital Speak: Part II

Picture yourself as an explorer in an ancient civilization. Before you could find work, you would first have to learn the language of the local tribe. After travelling to many villages, you would eventually find that everyone used similar tools — although they looked slightly different and were called different names. Little has changed in the past 200 or so years. Last fall, this column dared to ask the question — what makes our high-tech audio tools so hard to use? We're talking about digital audio work stations here. Wouldn't it be great if someone was foolish enough to go around to each tribe — I mean company — and translate their terminology into plain English. (Although someone was rumored to at tempt such a noble task some time ago, I heard they lost all of the information when their hard disk crashed.) We attempt to do just that, backing up our disk daily.

Rick Schwartz is a contributing editor to R•E•P and director of post-production at Music Animals, Los Angeles.

THE DAW DIALECT

We've all heard someone ask "what is 'such and such' called our your system." This article is intended to bridge-the-gap between different products. It is not intended to be a shoot-out or feature comparison. Think of it as a digital audio workstation glossary or language translator. Imagine trying to learn a foreign language such as Spanish without first knowing English. It's much easier to understand something if you have a familiar point of reference. With the premise: once you've learned one system, you've learned them all, let us continue on with our journey.

THE CATTLE CALL

Workstation manufacturers were faxed a brief questionnaire and encouraged to fill in as many blanks as possible. More than 20 manufacturers were contacted, including Akai, AMS, Digidesign, Doremi Labs, Hybrid Arts, Lexicon, Microtech, New England Digital, Otari, Roland, Sonic Solutions, SSL, Spectral Synthesis, Steinberg Jones, Studer EdiTech, Symetrix, Turtle Beach Systems and WaveFrame. (Unfortunately not everyone responded by press time, which is why some didn't make the chart.) Although we tried to include as many major, shipping systems as possible, we are aware that there are over 50 workstations rumored to be on the market. Needless to say, it didn't take long before our fax machine was ringing off the hook with responses.

Data was grouped into four main areas of interest: system conventions, general definitions, play commands and editing commands. The top section on the accompanying chart describes system conventions. Most programmers use a tape-based model with on-screen animation to simulate tape movement. Some systems move the tape, others move a simulated tape head. Not everyone rolls tape from left to right, depending on your perspective to the heads and the intended application.

PLAY TIME

Things are not as straightforward as they were in the days of the mechanical tape transport. In the good old days, when you pressed play, the machine started playing from the point the tape had last been stopped. These days it's not uncommon for a workstation to have up to 10 play commands. Because of their random-access nature, a hard disk can stop-on-a-dime. A workstation can play right up to an edit point and stop with single-sample accuracy. This feature is great for checking edits. Even if you catch a few milliseconds of the next beat, you

Description	Akai DD-1000	AMS Audiofile	Digidesign Pro Tools	Digidesign Sound Tools	Doremi Labs Dawn
System Conventions					
Indicates movement on the screen	play cursor	play head	counter	cursor	time line
Direction the screen moves	left to right	left to right	left to right	left to right	top to bottom
The recording media is called	MO drive	hard disk	disk	disk	rec. media
General Definitions					
Reference marker	GPM	mark	marker	marker	marker
A recording is called a	take	cue	sound file	sound file	sound file
Bundled audio data with edits	take	files	session file	sound file	edit list
A selected part of a recording	cut	segment	region	region	event
Play a specified region of a sound	play cut	play	play	play	play region
An item in a playlist	cue	event	region	region	cue or event
An EDL is called a	Q list	event list	playlist	playlist	event list
A graphic editing window	Q list edit	trim window	ProEDIT	editing window	mix window
Multiple versions of the same recording	renamed takes	takes	takes	copies	takes
Internal Storage Buffer	clipboard	cue library	region list	regions	clipboard
Play Commands					
Play sound before selected area	move GPM before	play up to	option left arrow	left arrow	play selection
Play sound between selected area	play cut	play	play	speaker icon	play selection
Play sound after selected area	move gpm after	play from	option right arrow	right arrow	play selection
Play sound across an edit	n/a	play across	option play	opt. cmd. speaker	play selection
Editing Commands					
Remove sound & shift all other sound to the left	define & assemble in Q List	insert segment	shuffle	cut region	bite & splice
Remove sound w/o changing sync (destructive)	cut & retake	overlay segment	delete (slip mode)	silence	clear
Remove sound & shift all to the left (destructive)	copy & select new cut	replace segment	n/a	n/a	delete
Remove part of a sound file & leave the rest	copy + T	spot segment	use slip mode	remove region	bite
Insert a sound at a start marker	copy + T	spot segment	paste	paste	insert
Insert a sound at an end marker	select new sound	spot segment	paste	paste	insert
Insert a sound immediately after another	insert in song mode	spot segment	drop region	paste	splice
Insert a sound, substituting one sound for another	drag & edit TC number	replace segment	paste	replace	place
Insert a sound and move all others to the right	block slip	n/a	shuffle	insert region	insert
Insert a sound at a specific time code location	slip	sync segment	use spot window	insert & set time	capture
Move a group of sounds in a playlist	change number while offline	block offset	shift select & move	shift select & move	offset selection
Move an element by one frame	zoom in	nudge	use spot window	adjust time code	bump
Resync an element while locked to timecode	n/a	n/a	n/a	n/a	capture
Sample level editingto match zero crossings	yes	n/a	use zoom tool	use zoom tool	auto-zero
To chase timecode you need to	set SMPTE as timesource	select external	go online	go online	ext. sync
To jump between markers is to	data +/-	nudge	hit autolocate #	type marker #'s	tab
To ungroup elements is to	exit block, slip page	offset	click outside	click outside	n/a

will hear a suction-type sound, instead of a clean stop. The ability to play across an edit gives you the ability to preview an edit without actually performing it. In addition, random-access capability allows the user to play any sound at any time code location instantly. Some work stations also give you the ability to play a sound file backwards, half-speed or twice play speed.

THE ELECTRONIC GREASE PEN

Before you can edit, you need to tell the system what to remove. Sound is marked using an electronic grease pencil. If your client changes direction during an editing session, just delete all of your marks and start over. The ability to label marks helps to keep things straight. It's possible to make very accurate marks on-the-fly. Once you've made your marks it's time to start cutting.

Actually, cutting is a misnomer, because most systems employ non-destructive editing. The system just plays around the cuts. Nothing is removed from the sound file. No matter how much you chop up a sound file, you can always go back to the original recording. Non-destructive editing is great, but there are times when you may want destructive editing as well. Let's say you're cutting dialogue for a western movie and a pager goes off on the set — you may want to remove that sound never to see the light of day again.

In addition, the disk space left after you cut out a file is available to be reused again (it seems like you can never have enough disk space available). There are several caveats to destructive editing, aside from the fact that it permanently changes your master recording. Unless the disk operating system rewrites the file, deleting many small chunks of sound will leave your disk looking like a piece of Swiss cheese. Disk fragmentation limits your maximum recording time and makes the disk work harder than it needs to. To avoid excess fragmentation, some systems will rewrite a new contiguous file on disk, which takes time.

SOUND PROCESSING

Digital audio workstations operate much like a word processor in the way they edit sound. Most editing commands are based on cut, copy and paste. Copy is self explanatory and was the only command everyone agreed upon, so I removed it from the chart to make more room. In the analog world, you copy or dub something like a chorus, so it could be used elsewhere in a song. Cut is similar to copy, except for the fact that it removes a selection from your original file. Unlike analog, you can cut out parts of a sound file without changing sync. Normally, sound after the edit moves to the left to fill in the hole.

Another difference between analog and disk-based digital is that once you cut or copy something, you can paste or insert it more than once. Sound can be inserted at a specific time code location or immediately after another. Digital editing allows the user to decide whether the next sound is ignored, replaced or shifted to the right. Cut and paste can be destructive or non-destructive in nature. Sound can also be moved simply without making a copy of it.

Some manufacturers included key commands or F-keys for their editing commands. It's important to strike a balance between command keys and menu commands. As companies add even more features, it will become increasingly more difficult to include everything in pull down menus, icon palettes or soft keys. Nested menus or command keys must be used. Using command keys is faster than pointing and clicking at the screen, but it takes time to memorize them. Ideally a system should work on both levels — it should be easy-to-use for the beginner, without being slow or clumsy for the experienced user.

Continued on page 64

Lexicon Opus	Micro Technology MicroSound	NED Post Pro	Otari Prodisk	Roland DM-80	Studer Editech Dyaxis 2+2	Turtle Beach 56k	Waveframe Audioframe
position indicator	play cursor	scrolling events	shuttle	scrolling	play head	play cursor	play head
left to right	R to L, L to R	right to left	L to R, T to Bottom	right to left	left to right	left to right	side to side
job/reels	disk	disk	disk	hard disk	disk	disk	disk
cue point	flag	marker	cue point	marker	log tic	marker	sync point
segment	sound file	cue	cue	take	sound file	sound file	recording
safe segments	mix files	events	n/a	project	view files	n/a	reel or track
section	segment	region	event	phrase	view	zone	event
section	play mark	audition	play	play-phrase	view file	zone audition	play event
segment	segment	event	cue	phrase	mix element	zone	cue or event
track list/file screen	mix file	EDL	cue list	playlist	track list	playlist	edit decision list
play/record screen	waveform window	Editview	cue editor	tape window	mix window	main editing screen	signal window
copies	user titles anything	takes	track versions	renamed phrases	takes	takes	re-takes
track	positional window	clipboards	clipboard	clipboard	clipboard	clipboard	clip board
play to edit	preroll	play to handle	cursor left	preview to	play before	play from	n/a
play	play mark	audition	shift click	play phrase	play	play sound	play
play from edit	post roll	play from handle	cursor right	preview after	play after	play from	end select, play
play edit	skip zone	n/a	n/a	preview thru	cut play	play across	n/a
track cut	mark zone, skip	delete	cut	cut	cut	playlist	close
segment cut	n/a	cut, replace cue	delete	n/a	cut, save selection as	n/a	replace
cut delete	n/a	n/a	delete & ripple	n/a	cut, save selection as	cut	close
cut insert	mark zone, silence	cut	erase	erase	cut	mute	replace
copy insert	snap to grid	paste or insert	place in	move	paste to left cursor	paste	lay-in
replace	snap to grid	paste or insert	place out	move	n/a	butt to zone	replace, split or lay-in
cut insert	Alt. S +A	paste	chain	move to marker	abut	paste	replace, split or lay-in
replace	n/a	exchange	delete/place	take charge	paste, cut	paste fill	replace or lay-in
insert	move all	fill	ripple	insert	n/a	paste	open
insert	snap to grid	place	place record time	move	paste to time code	n/a	replace, split or lay-in
track align	control click & drag	move	ripple all	insert	nudge or cut & paste	n/a	slip
align	drag to grid	slip	nudge	offset a move	bump	n/a	slip
n/a	n/a	align	n/a	yes	slew	n/a	split, slip
n/a	yes	trim	n/a	offset edit	fine locate	n/a	n/a
enable time code	enable chase lock	set ext. sync	chase on	set timebase to SMPTE	chase time code	n/a	select chase
go to cue point	click go to icon...	locate	click on	next/previous	tab	marker tabs	click on time in
n/a	unlock	de-select	n/a	split	deselect	n/a	n/a

THE R•E•P INTERVIEW:

STEVE LILLYWHITE

By Dan Levitin

British Producer/Engineer Steve Lillywhite is best known for the drum sound from hell — the huge reverb and compressed kick-snare sound that rode with Big Country and U2 to the top of the international charts. His sounds have been often imitated but never copied. On the release of his newest production — wife Kirsty MacColl's second Charisma release, "Electric Landlady" — Lillywhite took time out to speak with R•E•P about the album specifically, his approach to production in general and his feelings about rap music.

R•E•P: I like Kirsty's new record a lot. It struck in me the same reaction as when I first heard The Smiths — alternative music with these light, bouncy guitars...
SL: Great! That's exactly what we tried to do.

R•E•P: The album seems a bit of a departure from your other work. Compared to Big Country, U2, The Furs and so on, this is lighter.
SL: Yes, well, personally I've gotten off the big drum sound, which is something I used to do in the mid-'80s.

R•E•P: A lot of people say that you are responsible for having introduced that big drum sound to the airwaves in the '80s. The Steve Lillywhite drum sound that began with these alternative bands became mainstream with artists such as Bruce Springsteen. It's been said that you changed the sound of radio in the '80s.
SL: My goodness, that's very nice to be credited with that. Well ... I'd been experimenting with ambience around the time of Siouxsie and the Banshees' first album, which is a long time ago, and then I suppose when Peter Gabriel's album came out, and that had "The Intruder" song and stuff like that, we were using a lot of compression on ambience. Really, that's all it was.

R•E•P: And the snare was huge. It was drums from hell!
SL: That's right — we like that. At least we did in those days. The reason I don't like doing it so much these days is that everyone has that drum sound in a microchip. You know how it is when you're always looking for something a little bit different. This [Kirsty MacColl] is what I've been doing lately. I prefer drummers not to play so hard now, and I don't mic them so widely.

R•E•P: So what is the new Steve Lillywhite creative inspiration these days?
SL: Oh, I'm not sure I've got any brain cells left! I don't know...I'm into music more. I'm learning more about music rather than sound. I suppose when I worked with David Byrne on "Rei Momo" that influenced me a lot.

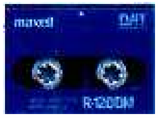
R•E•P: The musicians on that album were great, and you used them for some of the tracks on "Electric Landlady."
SL: Right. I really enjoyed the musicians who played on the "Rei Momo" album a lot, and so did Kirsty; it was her idea. She said, "Let's get those guys in to do some tracks on my album."

Dan Levitin is R•E•P's music production editor and a producer based in Stanford, CA.

STEVE LILLYWHITE PARTIAL DISCOGRAPHY

Big Country	"The Crossing," "Steeltown"
David Byrne	"Rei Momo"
The La's	"The La's"
Peter Gabriel	"Peter Gabriel 3"
Kirsty MacColl	"Kite," "Electric Landlady"
Psychedelic Furs	"Psychedelic Furs," "Talk Talk Talk"
Rolling Stones	"Dirty Work"
Simple Minds	"Sparkle in the Rain"
Siouxsie and The Banshees	"The Scream"
The Smiths	"Ask" (track)
Trash Can Sinatras	(In production)
U2	"Boy," "October," "War," "Joshua Tree" (remixes)
Ultravox	"Ultravox," "Ha Ha Ha"
XTC	"Drums and Wires," "Black Sea"

"Mimicking A Time-Honored Celebrity, The Fostex Portable DAT Became An Overnight Sensation."



The portable Fostex PD-2's advanced DAT technology lets you stay in the digital domain from location start to post production finish.

The back-lit LCD shows all function data, including internally generated SMPTE timecode and A-Time, plus levels, program number, battery condition and error indications.

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It features a built-in timecode generator/reader with jam sync capability. Word and composite video sync let you lock to vital external references. It has I/O connectors for both +4dBm analog as well as AES/EBU digital signals. You can choose T-12 or 48V phantom powering. Select any of four filters, a limiter or phase reverse.

Our four-head design gives you off-the-tape confidence monitoring. There's a built-in speaker and the headphone monitor can be switched to mono, A/B or stereo—even M/S stereo with our built-in decoder.

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FOSTEX

Circle (13) on Rapid Facts Card

R•E•P: And Johnny Marr and Elliot Randall play brilliantly.
SL: Well, Elliot was a real find, because he now lives in London. He's married to this English girl, and no one knows he's around.

R•E•P: What former work of his was it that stands out in your mind?
SL: I suppose the Steely Dan stuff.

R•E•P: Yes, he plays that solo on "Reelin' in the Years." Jimmy Page once called it his all-time favorite guitar solo.
SL: On one hand, I kind of want to keep him to myself, but on the other he's such a good player, I think he should play with everybody. He's great.

R•E•P: He *has* played with everybody — Paul McCartney's Ram, The Blues Brothers, Sha Na Na, Chuck Berry ...
SL: Yes, he's great. Johnny, too, of course, although Johnny Marr and Elliot weren't ever together at the same time on this album.

R•E•P: The song "My Affair" is a wonderful combination of a sort of "Copacabana" type of groove with a Smiths or XTC approach to songwriting.
SL: She co-wrote that with Mark Nevin, who is well-known for the Fairground Attraction. He also co-wrote some of Morrisey's new album.

R•E•P: There's tons of music going on in "Electric Landlady." How much of that was the producer, and how much was the songwriters and musicians?
SL: The game plan for the album was to record with as many musicians as it would take to make the backing tracks sound like a record, which was actually the game plan for "Kite" (MacColl's previous album) as well. But on "Electric Landlady" we went further, culminating in a session at Electric Lady studios in New York. For "My Affair" sessions, we had eight players at the same time, which is why you get this feeling of players playing off each other. It really was eight players all interacting.

We have a studio at our house, with a Sony/MCI 24-track, and so we just took the tapes and did all the vocals at home. Then we sat back and thought about what else we needed on it. In most cases, we found we didn't really need much more.

R•E•P: And what percentage of your input is musical?
SL: Well, it's hard to say. I'm not the kind of person to tell a bunch of Latin players what notes to play, though. I get in the right people, the people I know will play what I want to hear.

R•E•P: I guess everybody asks you this in interviews, but...
SL: No, I don't do many interviews at all. I really don't like doing interviews. Most of 'em make me nervous.

R•E•P: What unique problems are there producing your wife?
SL: Well, one of the big advantages is you get to sleep with her afterwards.

R•E•P: I suppose if you had a fight in the studio and then had to sleep with her afterwards, that might not be an advantage.
SL: Funny enough, we don't fight too much in the studio; she is a talented girl. She has some great ideas, and I just let the ideas flow.

R•E•P: Tell me about the miking for the album.
SL: Everything was pretty much close miked. I used an AKG 414 on her vocals up until the very end when I borrowed a TLM 170.

R•E•P: How did you like that?
SL: The TLM was brilliant! I wished I'd had it all the way through. Normally I have to EQ Kirsty's voice on a 414, to add top end because she's very quiet, but on the TLM I didn't need to do that. It had a crisper top end.

R•E•P: Because it's more sensitive to a soft voice you think, or because of its EQ curve?
SL: I think because of the sensitivity, yes. I was really very impressed. I have been using it a lot recently.

We were all young, you know, and felt like "Yeah, let's go for it!"

R•E•P: You're an engineer yourself, and yet lately you've been hiring engineers.
SL: I used to be an engineer, but I've taught myself not to look at the meters. I like sitting at the desk, and I like to balance [mix], but I don't really like getting sounds too much. I do like to get the echoes and balance the monitors. I'm not really that confident as an engineer anymore. The engineer who did Kirsty's record, Jonn Fausty, is very, very good. He's engineered something like 3,000 Latin albums. He's a free-lancer in New York.

R•E•P: What else can I ask you about Kirsty's album?
SL: I don't know, let me think...it is a bit of a departure for me, I suppose. I'm quite pleased with the sound of it. It's not my sort of sound, really ... it's a bit wider sound. You can hear things more clearly than in my other records.

R•E•P: You make an interesting point. Your U2 albums ...
SL: They all sounded a bit dense, I think.

R•E•P: Yes, it's a funny thing. They were very influential, and they're very powerful, but from a technical standpoint, they sound terrible.
SL: Yes, they're dreadful, aren't they?

R•E•P: And yet there's some genius in them. The drums are too loud, the bass is too loud — everything sounds too loud, and too wild, too dense. It was a very bold way to record. It established, I think, a whole new sound and way of using the studio.
SL: Well, we were all young, you know, and felt like "Yeah, let's go for it!" It sounds much like a band playing live, but it wasn't done like that. There were a lot of overdubs and layering and stacking up of sounds.

R•E•P: I think the genius in the recording, and why everyone is so enamored of the sound, is because it sounds like some maniac just stuck up a couple of mics at random and recorded some wild band live, while the sound guy was on acid and making a lot of wrong moves.
SL: (Laughs) I don't know. I've never done a mix on acid, so I don't know. I did three albums with U2, and they never did a song live before recording it after the first album. "October" and "War" were written entirely in the studio. After the recording was done and I went to hear them live, I thought they sounded better.

R•E•P: Where did you come up with the concept for recording them the way you did?
SL: The drum sound came about because the sound I wanted had to be done out in the hallway at night. It had to be nighttime because during the day there was a girl at the end of the hall answering the telephone.

R•E•P: Your career has followed an interesting path in that, with the exception of the Rolling Stones record you produced, everything you've worked on is alternative music. They are groups and albums who have strong and devoted followings, but for the most part, they are not Top 40 smashes, they don't sell into the mutli-platinum category. They are influential albums, though, the kind that other musicians and producers hear and are influenced by. Why haven't you become involved with more mainstream artists?
SL: Because there aren't many mainstream artists that I like. There is one, R.E.M., that I would die to work with. They're great artistic people. But, I mean, who is there? Who would I work with? Bruce Springsteen!? I would have liked to work with him when he was younger, earlier in his career.

R•E•P: Vanilla Ice!
SL: No! Why? What would be the point?

R•E•P: I don't know — you could make a ton of money.
SL: I don't want to make a ton of money. I want to make music I like, and to work with people I like. Money! How much money can you spend? God ...

R•E•P: Will you ever make a rap album?
SL: Well, there's a rap on "Walking Down Madison," but I wouldn't want to make a whole rap album because it's such a limited musical concept. I'm aware that it's good, and I like some rap records, and I'd like to maybe use it in something, but not a whole album. It would get boring, wouldn't it?

I've always just done artists and people that I like, with the exception of the Rolling Stones, because you can't say "no" to the Rolling Stones. It's just a pisser that it was at a time when Keith and Mick weren't friendly with each other.

R•E•P: Well, that's a crap shoot.
SL: So there you go. It would have been nice to work with them when they were younger, but even so, it's nice to know you've done a Rolling Stones record.

R•E•P: Yeah, it kind of fits nicely into one's discography.
SL: I don't have a discography. Or did you say earlier you have something that has all the things I've worked on?

R•E•P: Yes, your management gave it to me. Would you like me to give it to you?
SL: No, that's OK, I'll try to get one from them. ■

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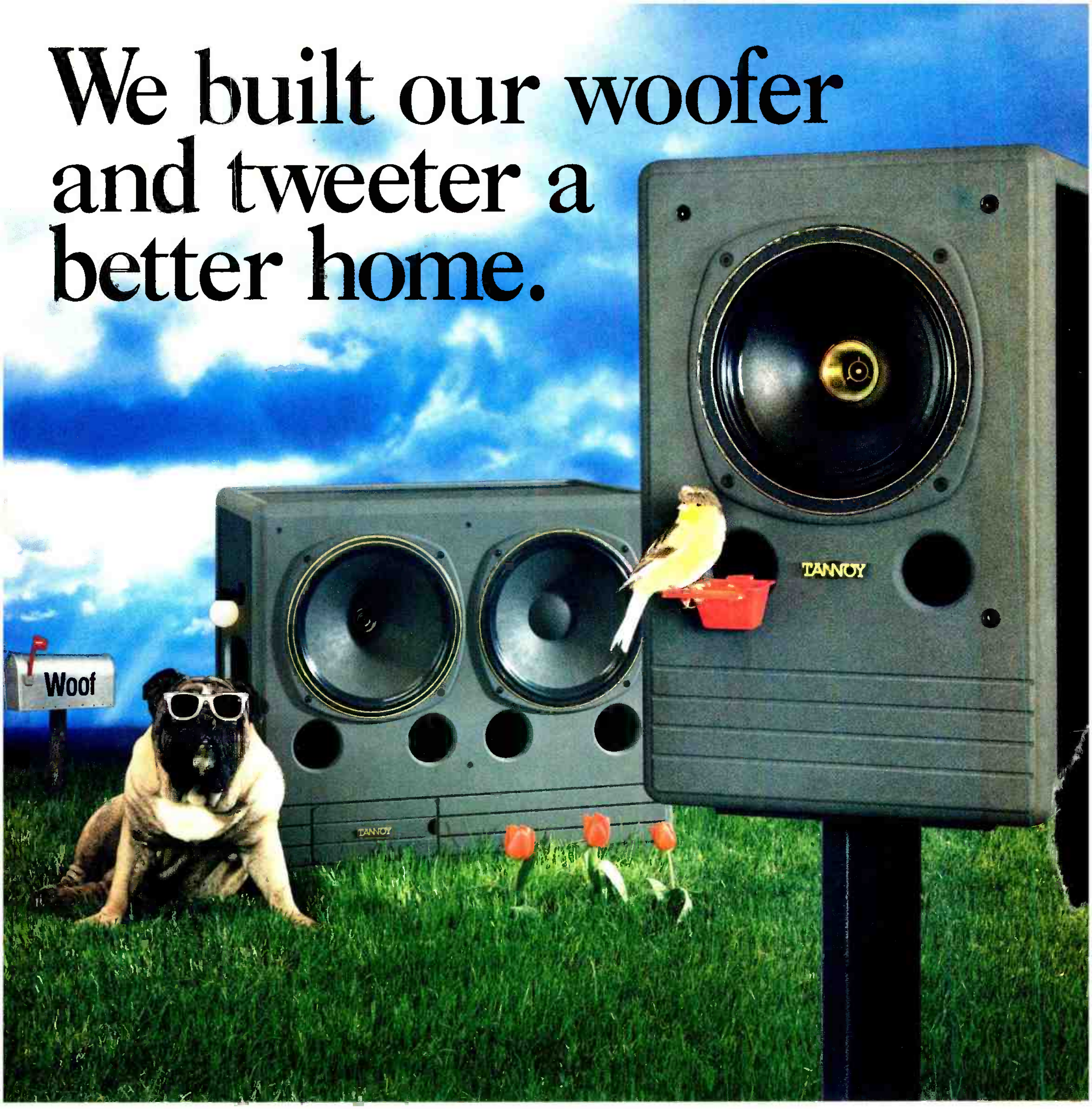

DMT AND TRANSDUCER HOUSINGS —

All Cabinets used in the new Tannoy Monitor Series represent considerably more thought and design than the average 'chipboard cabinet of the correct volume'.

Cabinets are constructed from a high density space-frame with rounded corners and edges, supporting MDF/high pressure twin laminated walls.

Rounded corners and edges greatly inhibit sound reflections and diffractions from cabinet boundaries. These can be major sources of irregularities in the reproduced sound, particularly in terms of the perceived placement of instruments within the sound stage. For the high frequency unit to make an effective job of launching all the detail that it can generate into the listening space, it must be held rigidly in the cabinet throughout its operational frequencies.

This is the key to the Tannoy cabinets. If a 'rigid' cabinet is used, the redundant energy from the rear of the bass unit and frame cause endless resonance problems within the cabinet. Differential Material Technology provides the answers by using a variety of different adhesives between the rear of the drive unit and brace, the cabinet walls and the brace and within the layers of the MDF laminate.

The lossy couplings effectively transmit and absorb energy in a frequency selective way. Put more simply, at low frequencies the drive unit sees the cabinet as a rigid structure and at higher frequencies as a resonance absorbing/damping structure.

DMT provides an ideal cabinet solution for the complete frequency range, eliminating unwanted loudspeaker biases.

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Circle (14) on Rapid Facts Card

REPLACING THE CABLE

By Anthony McLean

Some day, archaeologists digging down to our century will declare the 1990's to be the end of transitional analog technologies. Slightly deeper, the spoon diggers will unearth the fact that Bell Labs formatted audio signal transmission down copper cable by defining decibels and 600Ω lines in the 1920's. They will undoubtedly observe and chronicle analog's slow drift toward extinction. And they will surely notice that sometime in the 21st century planet earth went totally digital.

Fiber optic systems have traditionally been Fortune 500 domain; AT&T territory. Phil Lambert, a Siemens fiber optic systems engineer at their Princeton, IN, Potter and Brumfeild plant, emphasizes that fiber optics' big work is to link data networks and video and facility controls such as climate and security. Michael Pascazi of Fiber Optek Interconnect in Wappingers Falls, NY, believes the relatively tiny professional audio market hasn't held the same allure for vendors as seven-figure fiber optic installations. Audio, it seems, is the least common fiber optic application and the smallest piece of the fiber optic pie.

Anthony McLean is features editor of R•E•P.

AT&T Network Cable Systems

DIG THIS

Causing no capacitance, no inductance, and no hum, multiplexed digital signal transfer is clearly superior to analog. For simple audio signals, even twisted pair copper wire can be used to transfer digital audio. But at the high end, fiber optics easily manage audio and video signal, plus all other production-required data exchanges such as SMPTE, MIDI and RS-232.

Eventually, audio workstations (a.k.a. digital production workplaces) will force universal transition to bi-directional fiber optics within the studio production environment. In 1992, audio engineering standards and practices are surfing the second wave of digital technology. Near term, digital transmission interface seems

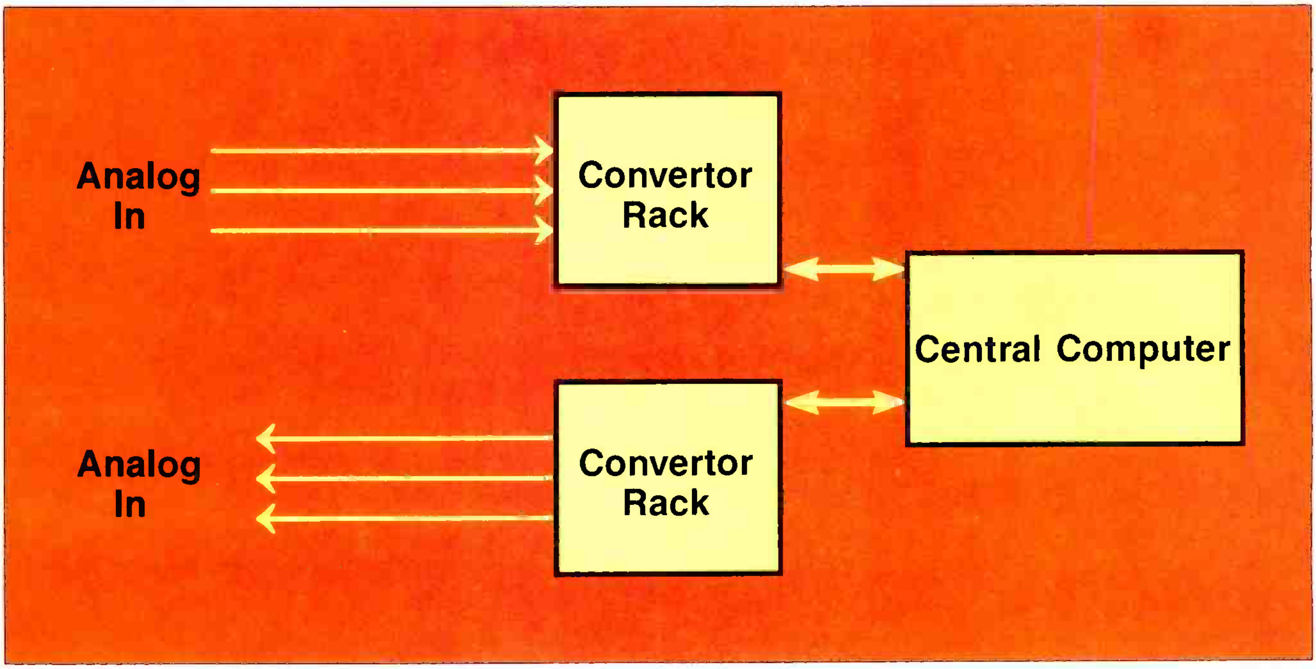

Figure 1. Centralized Fiber Optic Processing Topology.

finally positioned to penetrate professional audio. It's the current wave. Fiber optics, arguably the most elegant of transmission formats, are up ahead on the beach.

Since early fiber optic cabling was relatively brittle, live sound applications were out of the question. Expensive A/D and D/A conversion inhibited fiber optic penetration into production suite environments.

But as chip-based digital technology has improved and optic cable production capacity has skyrocketed, fiber optic hardware now approaches justifiable performance and expense territory for professional audio applications. In the '90s, the question changes from "if" to "when" pro audio will go fiber.

The fiber optic lexicon includes familiar terms such as transmitter/receivers (a.k.a. data links), transceivers, relays, interface modules and connecter/jumper hardware.

Giants such as Siemens International, in an effort to cross over from phone to commercial audio, manufacture components meeting FDDI (Fiber Distributed Data Interface) standards. While unsuited to professional audio, FDDI successful applications include CCTV systems in Munich's new airport and mega scale computer networking. Other standards, such as the T1 and T3 telephone standard also fail to reach professional audio's requirements.

GLASS/COPPER

A single optical fiber, or twisted pair, can simultaneously transfer various data types (its all data to the cable) via digital multiplexing. This encode-decode process integrates multiple audio and other inputs into a hyper-speed serial data stream output which is essentially loss free and signal phase/flat out to gigahertz.

The serial data string is primarily dependent only on its relative timing location to be accurately captured and processed.

As such, the key to any fiber optic data exchange is the system's fail-proof capacity to encode and decode, via timing control, all of the signal data (digital information) within the data stream.

Two basic approaches to fiber optic systems have evolved. Central processing, which is common to telco systems, is the original and most common application topology. Central processing strategy consolidates sub-systems, concentrates hardware and is perfectly matched to the telco universe. (See Figure 1.)

Less common is distributed processing topology. This approach, which is analogous to multiple outboard racks of routing, distribution and digital signal processing, seems well suited to professional audio. In general, distributed processing emphasizes minimum lengths of analog signal paths by making A/D conversions in relatively close proximity to the source and maintaining digital signal and processing control for as long as possible.

In the event of component or cable failure, the distributed technique can immediately construct an alternative signal path through application of a fail-safe switching program. While differing according to vendor, distributed processing can automatically execute glitch-free, hyper-fast bypass and reconnect commands. (See Figure 2.)

The marriage of fiber optics and audio production seems long overdue.

Since signal strength and risk management (i.e. keeping the signal flowing) constitute pro audio's universal "Job #1," the marriage of fiber optics and audio production seems long overdue.

THE LONE STAR CONNECTION

Lester Audio Labs of Dallas expects to deliver their first audio-fiber commercial system in January 1992. Named the DAS 2000, the rack-mount package fits eight channels per card to execute the A/D/A signal conversion, and is expandable to 64 send and 16 return channels.

Lester's software-based "soft patch" control uses an expandable 32×32 matrix for multiple routing, I/O, gain staging, and phantom power configurations.

Non-volatile memory stores 190 patch configurations for point-to-point or point-to-multipoint switching.

Designed for mobile applications such as broadcast remote trucks and touring sound reinforcement, Lester beta-tested their system with Turner broadcasting at the 1990 Seattle Goodwill games.

Paul Trimble is a veteran sound engineer and Lester's director of new product development. As the portable systems begin to ship, Trimble is developing fixed installation systems for new construction and retrofit.

Acknowledging the global production market, Trimble noted that, "The broadcast situation, the sound reinforcement situation, and studio installation, as well as theme-park/hotel complexes are all areas where multiple stages to some master control and multiple output points," define a potentially large fiber optic user base.

Acknowledging that "smart buildings, completely interactive, hold the greatest international market potential," Barry Thorton of OptoDigital in Austin, TX, anticipates an immediate $1.15 trillion annual market in fiber optic broadcast sales for improvements and capital expenditures.

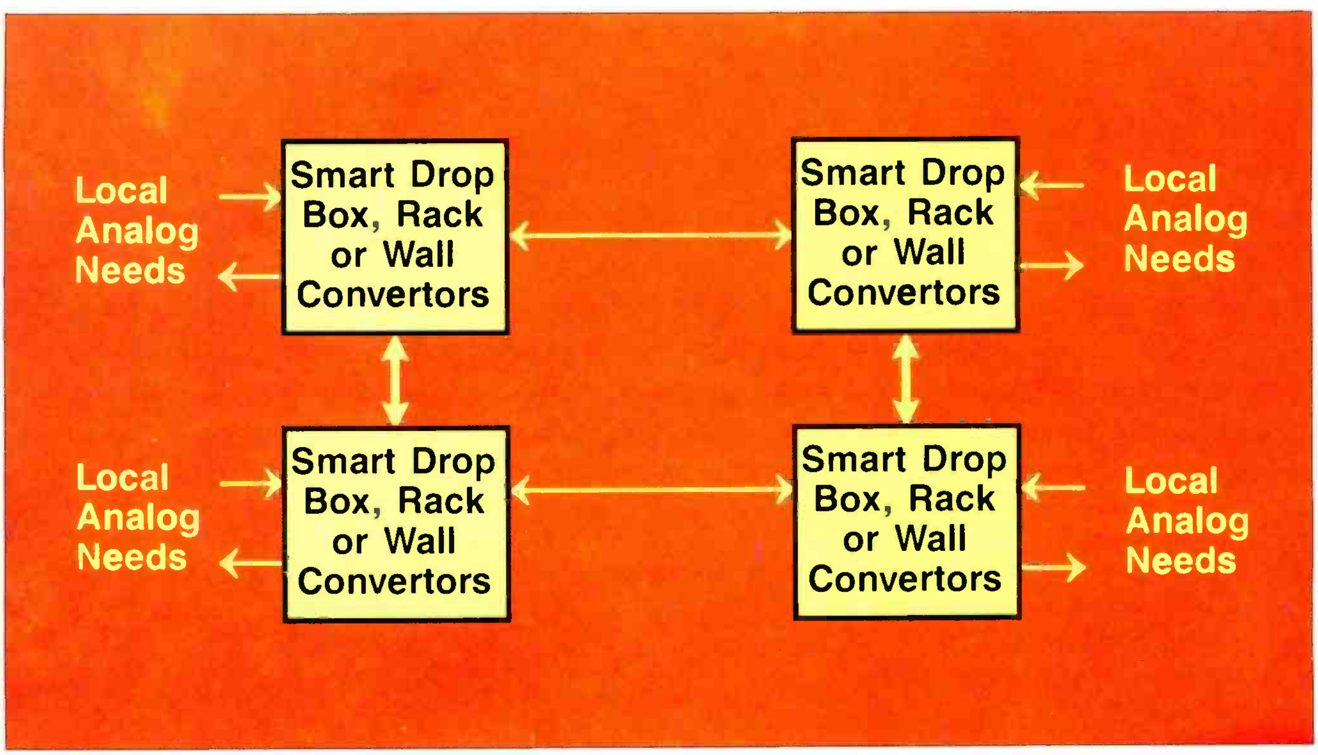

Figure 2. Distributed Fiber Optic Processing Topology.

Particularly strong is the state-supported European broadcast market. Continental governments use TV as a sociological tool, and don't hesitate to spend on transmission technology. Specified for all new Spanish TV installations, OptoDigital has also equipped many European video remote trucks.

TO NODE OR NOT TO NODE

For professional audio, the less time and distance that audio signal remains as analog the better. The fewest possible D/A and A/D conversions are also best, so OptoDigital recommends A/D conversion near the source. Once digital, all signals are merged and multiplexed at a device OptoDigital calls their "Node.

Nodes act as the ultimate electronic traffic cop. Essentially, a node is just smart enough to do its specific job, but lacks the knowledge to run the entire system. Some nodes simply switch and route. More intelligent nodes, after receiving electrical digital signal output from the A/D converters, output in optical, and simultaneously multiplex, switch and route within proprietary Bendix connecters.

Transition to fiber optic is, according to Thorton, "A migration path from the old to the new, not a stampede." Thorton added that,"Everyone agrees that by pulling the fiber you wont get hurt. Pick some small, almost irrelevant, part of your business ... put the fiber in there. When (users) forget it's there, it is assimilated."

OptoDigital has petitioned the EIA to have RS 9000 standardized for protocol and cable pin-outs. Reasoning that a 450Mbyte standard is needed for professional performance and to facilitate global interface, OptoDigital's goal is to achieve some common ground for the "migration" to fiber optic practices. While some structures, such as Hewlett Packard's HPPI exceed the 450Mbyte rate, (up to 622Mbytes) none have achieved a true "default" status.

One seductive byproduct of OptoDigital's system is the DSP resident in its A/D converters For users needing interactive, temporal control of multiple signal paths, the possibilities loom large.

Signal distribution reliabilities are, in a high risk business such as live broadcast or performance, serious as a heart attack. OptoDigital's routing and processing control, as applied to risk management, is designed for "can't-ever-fail" venues. The alternative path controls inherent to their software driven fiber optics are intended to eliminate potential interconnect failure. Redundancy is the order of the fiber optic universe, and OptoDigital understands that clearly.

ANOTHER PLAYER

Michael Creamer, vice president of sales/marketing for Bec Technologies agrees that, "There are no second chances!" Interface failure of a remote truck at a Stones pay-for-view, or a Holyfield-Tyson fight, could ruin the company at fault.

Bec, with offices in Orlando and Seattle, came to the audio industry from the military/industrial complex. Bob Proctor, vice president of engineering, produced his first prototype 40-channel multiplexed, wired-based system in 1988.

Initially, Bec systems were large format (64- to 80-channel) mainframe configurations. Internal market research concluded this was not what the market desired. Bec then redesigned its twisted pair and fiber systems so that modular components could be strategically assembled for various sized applications.

Bec's current hardware is fifth generation and uses an AT&T drive chip set. The 5-device system can be assembled in multiple configurations, features dc backup and is anchored by a programmable logic device Bec calls its PLD. Easy pin-to-pin chip set changeout is inherent to Bec's upgradable design.

Using three levels of redundant communication, Bec seeks to eliminate risk through its Fault Tolerant Redundant Communications. According to their marketing statement, "Fault tolerance uses redundant systems to monitor performance, switch any faulty piece out of the system, bring one of the backups on-line and let the engineer see that a failure has occurred so that he can service the unit at a more convenient time." Bec's system synchronizes on every sample to remain fault tolerant.

Redundancy is the order of the fiber optic universe.

Creamer uses the analogy of "a light pipe with taps" to communicate fiber optic signal circuitry. With the equivalent of a file server at each tap location, Creamer insists fiber optics is simply a better mouse trap. When asked about fiber optic's obvious utility within a digital environment Creamer responded, "I don't think it's so much the digital, it is just a better way to transmit."

OLD DOGS, NEW TRICKS

"On the horizon," Creamer says, "is a generation of (performance) consoles with mic preamps and A/D conversion taking place near the mics." But for various reasons, audio vendors seem reluctant to adopt digital transmission technologies.

At the big picture level, Creamer feels digital signal transmission eventually "... will change the way people do things." By removing time consuming and gut wrenching signal distribution hassle, Creamer envisions a time when "... (engineers) can spend their time on more important things, like the creative process of mixing."

Sudden impact of a change to fiber optics in drive line technology would be profound. If, and when, touring technologies assimilate digital signal transmission, distro gurus will likely change status from drop-dead important to nonessential commodities.

Factor the reality of a 1- to 3-D/A signal split without signal loss, add the fact that a fiber optic cable splice can be executed in five minutes, and the future seems clear. Fiber optics are definitely the audio snake and splitter for all of us.

PUSHING THE ENVELOPE IN THE REAL WORLD

Craig Dory spent six years in strategic planning and exploration development at Bell Labs. Eventually, Dory left Bell Labs to merge his fiber optic experience and self-educated recording chops.

The result, Dorian Recordings, has redefined classical music recording. Dorian customizes nearly all of their gear. Using Steven Paul-modified microphones, mic specific Jensen preamps and Wadia DigiLink A/D converters and fiber optics, Dorian has established an entire environment appropriate to digital recording.

These electronics comprise what Dorian calls its "pod". A Dorian pod integrates non-phantom (direct voltage to mic element) powered preamps and highly evolved A/D converters with special dithering circuitry. Close (within 15 feet) coupling between pods and transducers minimizes inherent analog signal problems such as capacitance, inductance and temporal deviations. Optical output maintains this pristine signal until its destiny with a DAT recorder.

Dorian records globally, but uses the acoustically renowned, 1,200-seat, Troy Savings Bank Music Hall as home. Labor intensive preproduction earmarks all Dorian recordings. Foregoing equalization, the Dorian technique places a premium on input signal, with mic placement as the ultimate detail. Nothing gets fixed in the mix, because nothing is left to fix.

Dorian assures success, via redundancy, by simultaneously running tape on as many as four recorders, each with a particular mic array. The results are nothing short of spectacular.

Dory candidly points out that, "I was surprised. (When You) synergize all your electronics, and eliminate mic cable, even at the 50-foot level, it's definitely worth it." Dorians' modifications aren't for sale, but Dory did speculate that his approach might transfer to a bigger production universe, depending on budget.

Obviously, this is audiophile stuff, but one wonders what 48 digital tracks of this would sound like?

LETS GET META-PHYSICAL

Marshall McLuhan distilled the future into his conceptual world village and the production environment is moving, via hyper-media upgrades, to online status with whole earth scope. McLuhan could not, however, have imagined how his concept would become say ... virtual reality.

When the inevitable happens, and universal fiber optic interconnect between our production facilities occurs, the effect will transcend mere production.

It will accelerate the evolution to a world village. Creative communities, such as your favorite studio, will assume the role of international artist colonies. By eliminating distinctions between video, audio and allied production technologies, digital interconnect will sledgehammer the way we do our work. ■

FOR FURTHER INFORMATION

Lester Audio Labs
214-637-9311; fax 214-637-9314.

OptoDigital
512-338-4707; fax 512-794-9997.

Bec Technologies
407-855-8181; fax 407-856-7516.

The Dynamic Duo

TC 2290: Frequency Response: 20-20KHz, +0 / -0.5 dB
THD: < 0.05% 1kHz, 0 dBm; Dynamic Range: >100 dB
Digital Conversion: Dynamic Differential
Sampling Rate: 1MHz; Max. Input Level: +22dBm

M5000: Frequency Response: 10-22KHz, -0.5 dB, Fs=43KHz
THD: < 0.01% 1kHz, 0 dBm; Dynamic Range: >100 dB
Digital Conversion: Linear 18bit 64X in, 20bit out
Sampling Rate: 48KHz, 44.1KHz, 32KHz; Max. Input Level: +22 dBm

The legendary TC2290 has a new partner! Meet the M5000 Digital Audio Mainframe, the world's first user expandable DSP system. You've relied upon the TC2290 for the ultimate in digital delay effects for years, now the M5000 is the latest tool for Reverb, Ambience and Pitch Shift effects. Like the TC2290, the performance of the M5000 is simply astonishing. In fact, the M5000 will change forever the way you think about moderately priced digital signal processing. Why? The M5000 uses T.C.'s proprietary **DARC** (Digital Audio Reverb Co-processor) technology. **DARC** boosts the M5000's digital processor well beyond normal levels of performance and provides the power to support the complex algorithms desired by even the most demanding audio professional.

The M5000's High Speed 24bit buss supports up to four modules. The standard M5000 configuraton of 1 AD-DA module and 1 DSP module leaves an additional pair of empty slots. Want two completely independent true Stereo processors with Analog and Digital I/O in two rack spaces? Simply add a second set of modules for a fraction of the cost of an addtional unit! All Digital Studio? Configure the M5000 accordingly with up to Eight channels of Digital Processing with AES/EBU, SPDIF and Optical I/O. Standard Interfaces include MIDI, Ram Card and SMPTE (in). Options include SCSI, LAN, & Floppy Disc Interfaces. Future M5000 modules will allow you to upgrade your unit as technology evolves. Say good-bye to planned obsolescence!

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Circle (17) on Rapid Facts Card

A 1992 analysis of the important professional features today's DAT recorders provide.

DAT ANALYSIS

By Rick Schwartz

Even though most of today's professional DAT machines seem to offer similar features and capabilities, one almost has to look past the spec sheets and feature sets to find out what they really provide, functionally. We've done the footwork for you.

Here's an up-to-date analysis of the current crop of DAT recorders. We hope this helps you in your shopping and spec-ing activities.

AIWA HH-B1

A popular new portable from AIWA (with help from the U.K.'s Hilton Hire) uses 1-bit A/D converters. The HH-B1 supports both AES/EBU (via an XLR splitter lead) and IEC digital formats. SCMS can be avoided by using the AES outputs. According to the manufacturer's literature, the unit will not record at 44.1kHz using the analog inputs; weighs only 0.9kg (without batteries); will record for up to 220 minutes using AA dry cell and lead-acid rechargeable batteries; has the distinction of being the only portable that can write, erase and renumber program IDs; and also will record absolute time code.

FOSTEX D-20

This unit achieved popularity by being the first DAT on the market with SMPTE time code capabilities. The D-20 reads and writes SMPTE time code, which conforms to the AES/EBU standard, as well as the Fostex time code standard. It has word-clock inputs and outputs. The D-20 is a 4-head machine with monitoring off-tape during recording. Because the unit has been around for awhile it escaped SCMS. Indexes are not transferred during digital dubs, because the unit only has an AES-EBU digital port. It has no auto indexing capability. Post-striping SMPTE time code is useful, but program IDs are erased in the process. Although the unit will perform audio punch-ins, they are not always glitch free. The D-20 is still the only machine on the market with pitch-control, although the process creates problems on digital transfers.

Rick Schwartz is a contributing editor to R•E•P and director of post-production at Music Animals, Los Angeles.

JVC DS-DT900

The largest DAT player on the market, it has 64× oversampling A/D converters with 3-stage FIR digital filter and fourth order noise shaper; reads and writes SMPTE time code and will slave to external sync; has an hour meter that indicates tape head wear. Also features a built-in head cleaner; has AES/EBU digital interface only; will perform digital fades; and may be remote controlled via a 45-pin parallel interface, or a 9-pin serial interface.

PANASONIC SV-255

This is Panasonic's popular portable with lower noise specs and better-sounding filters than its predecessor — the SV-250. The unit weighs only 3.2 pounds and has a rechargeable battery that will last up to 2.2 hours; has a unique dual-channel mono mode that records one channel −15dB lower to protect from signal overload; uses a non-standard unbalanced connecter for S/PDIF digital output. Because it was designed to be used in the field, the unit has no digital input, and it will not record at a sampling frequency of 44.1kHz; does not allow the user to write or erase program IDs or renumber start IDs; program IDs are not transferred during a digital dub; and the unit does not read or record absolute time.

PANASONIC SV-3500

Although this is an older machine, we include it for reference purposes because of its use in so many professional facilities. It will copy protected CDs, by changing an internal switch; does not acknowledge SCMS, so a user can make unlimited digital copies; and program IDs are not transferred during a digital dub. Head cleaning is extremely difficult. To properly clean the heads the entire unit has to be disassembled, which can take several hours. It also includes a selectable error rate readout.

PANASONIC SV-3700/3900

Except for their remote controls and the fact that the 3900 has a RS-422 port, both of these machines are similar, using same transport and one-bit converter technology. Both use the IEC-958 format for consumer digital I/O. Minor digital transfer problems caused by the new format can be solved by cutting a capacitor inside the unit, which voids warranty. Contrary to popular belief, copy-prohibit is completely avoidable on both consumer and professional digital inputs. Copy prohibit bits can be ignored, passed on, added or removed although control over SCMS would be easier if the back panel DIP switches were labeled. As long as there is ample time in-between indexes, the consumer ports will transfer program IDs during a digital transfer. Panasonic is one of the only manufacturers that post-records absolute time code. Both machines write AES/EBU standard time code that can be read on any SMPTE time code machine. The SV-3900 will locate to absolute time using optional Macintosh software. However, without purchasing the optional remote for the SV-3900, a user cannot access several important features, including end search. Ironically, several essential features are missing from the remote, including auto record mute and door open. It has a built-in error rate readout display for both heads.

SHARP FXD200/YAMAHA DTR-2

Some people feel this machine is one of the best-sounding units on the market, with its dual 1-bit A/D and D/A converters with oversampling digital filters. Although the unit has fiber optic and coaxial digital I/Os (using a locking industrial-type BNC connector) it doesn't support AES/EBU. Sharp does not acknowledge copy prohibit flags, so users can make unlimited digital copies. The Yamaha DTR-2 has balanced and unbalanced connectors. Both units will locate to absolute time location and both offer 32kHz extended record and play capabilities.

SONY PCM-2000

Although Sony's professional time code portable has been out for almost four years, the time code section hasn't worked effectively until recently. There is no way to pre-stripe or post-stripe a tape with SMPTE time code, which limits the device's usefulness in a post-production environment. It is one of the only units on the market that will record at 44.056kHz. It has only AES/EBU digital I/O; cannot write, erase or renumber program IDs, and will not record absolute time. Weighing in at 9 pounds 4 ounces (without the BVG-200 time code generator) the PCM-2000 is the heaviest portable on-the-market.

SONY PCM-2700

A much-improved version of the PCM-2500, this unit has confidence playback heads for read-after-write capability and 1-bit D/A converters, and will record and playback at 32kHz for extended play capabilities. Although the

(1) AES/EBU is accessable via a mini-connector and a splitter cable which is included
(2) Start IDs will be transfered during a digital dub if the S/PDIF port is used and AUTO is on
(3) Full Control over SCMS is provided via rear panel DIP switches
(4) The Panasonic SV-3900 will located to absolute time using optional Mac software
(5) Digital output appears on a 2.5mm miniture pin plug which requires included adapter cable. No digital input
(6) Uses a professional BNC connector
(7) Uses a 1/4-inch Jack with optional adapter cable
(8) New version of the TCD-D10 will read and write absolute code
(9) With an optional 30 pin cable
(10) Although there is no time code input, the units record SMPTE-compatable absolute timecode
N/A = Not Applicable

PROFESSIONAL DAT RECORDERS	Aiwa HHB1PRO	Fostex D-20	JVC DS-DT900	Panasonic SV-255	Panasonic SV-3500	Panasonic 3700/3900
Digital I/O:						
AES/EBU (XLR-3 type)	Yes (1)	Yes	Yes	No	No	Yes
IEC-958/Type-2 (RCA type)	No	No	No	No	No	Yes
S/P DIF (RCA type)	Yes	No	No	Yes (5)	Yes	No
OPTICAL (EIAJ CP-340)	No	No	No	No	No	No
Metering & Display						
Internal Metering Reference Level (+4 dBu)	-18dB	-18dB	-18dB	-18dB	-18dB	-18dB
Meter Range	-40 to 0	-50 to 0	-60 to 0	-50 to 0	-60 to 0	-60 to 0
Error Rate Readout	None	Yes	1 LED	Yes	Yes	Yes
Program IDs						
Will Renumber Program IDs	Yes	Yes	Yes	Yes	Yes	Yes
Manual Writing of Start/Skip IDs	Yes	Yes	Yes	No	Yes	Yes
Will Copy Program IDs During Digital Dub	No	Yes (9)	No	No	No	Yes (2)
Copy Protection						
Will Record from a Copy Prohibited CD	Yes	N/A	N/A	N/A	Yes	Yes
Adds SCMS Copycode to Digital Copies	Yes	No	No	No	No	Yes
Will Ignore SCMS Bit on S/PDIF Ports	No	N/A	N/A	N/A	Yes	Yes (3)
Will Remove SCMS Bit During Digital Dub	No	N/A	N/A	N/A	No	Yes
Time Code:						
Stripes Absolute Code While Recording	Yes	Yes	Yes	No	Yes	Yes
Reads Absolute Time Code	Yes	Yes	Yes	No	Yes	Yes
Post-stripes Absolute Time Code	No	No	No	No	Yes	Yes
Will Locate to Absolute Time	No	No	No	No	No	Yes (4)
Records SMPTE Time Code	No	Yes	Yes	No	No	No
Post-stripes SMPTE Time Code	No	Yes	No	No	No	No
General						
Portable	Yes	No	No	Yes	No	No
Confidence Head Playback	No	Yes	No	No	No	No
List Price	$2,000	$8,500	$4,500	$2,700	$2,500	$1600/$2100

(1) AES/EBU is accessable via a mini-connector and a splitter cable which is included
(2) Start IDs will be transfered during a digital dub if the S/PDIF port is used and AUTO is on
(3) Full Control over SCMS is provided via rear panel DIP switches
(4) The Panasonic SV-3900 will located to absolute time using optional Mac software
(5) Digital output appears on a 2.5mm miniture pin plug which requires included adapter cable. No digital input
(6) Uses a professional BNC connector
(7) Uses a 1/4-inch Jack with optional adapter cable
(8) New version of the TCD-D10 will read and write absolute code
(9) With an optional 30 pin cable
(10) Although there is no time code input, the units record SMPTE-compatable absolute timecode
N/A = Not Applicable

PROFESSIONAL DAT RECORDERS	Sharp SX-D200	Sony PCM-2000	Sony PCM-2700	Sony TCD-D10	Tascam DA-30	Technics DA-10	Yamaha DTR2
Digital I/O:							
AES/EBU (XLR-3 type)	No	Yes	Yes	Yes (7)	Yes	No	No
IEC-958/Type-2 (RCA type)	Yes (6)	No	Yes	No	No	Yes	Yes
S/P DIF (RCA type)	No	No	No	Yes (8)	Yes	No	No
OPTICAL (EIAJ CP-340)	Yes	No	No	No	No	No	Yes
Metering & Display							
Internal Metering Reference Level (+4 dBu)	-18dB	-20dB	-20dB	-20dB	-16dB	-18dB	-18dB
Meter Range	-50 to 0	-60 to 0	-60 to 0	-50 to 0	-50 to 0	-60 to 0	-50 to 0
Error Rate Readout	No	No	No	No	No	Yes	No
Program IDs							
Will Renumber Program IDs	Yes	No	Yes	No	Yes	Yes	Yes
Manual Writing of Start/Skip IDs	Yes	Yes	Yes	No	Yes	Yes	Yes
Will Copy Program IDs During Digital Dub	No	Yes	Yes	No	No	Yes	No
Copy Protection							
Will Record from a Copy Prohibited CD	Yes	N/A	Yes	No	Yes	Yes	Yes
Adds SCMS Copycode to Digital Copies	No	No	Yes	No	Yes	Yes	Yes
Will Ignore SCMS Bit on S/PDIF Ports	No	Yes	No	Yes	No	No	No
Will Remove SCMS Bit During Digital Dub	No	No	No	No	No	No	No
Time Code:							
Stripes Absolute Code While Recording	Yes	No	Yes	No (8)	Yes	Yes	Yes
Reads Absolute Time Code	Yes	No	Yes	No (8)	Yes	Yes	Yes
Post-stripes Absolute Time Code	No	No	No	No	No	Yes	No
Will Locate to Absolute Time	Yes	No	Yes	No	No	No	Yes
Records SMPTE Time Code	No	Yes	No	No	No	No	No
Post-stripes SMPTE Time Code	No	No	No	No	No	No	No
General							
Portable	No	Yes	No	Yes	No	No	No
Confidence Head Playback	No	No	Yes	No	No	No	No
List Price	$2,200	$8,520	$2,900	$2,900	$1,500	$1,000	$1,495

Continued from page 34
unit will accept signals from an AES/EBU source, signal level adjustment may be required. By entering an absolute time address via numeric keys, location to any point on tape can be achieved. It has a rehearsal function that allows the user to trim start, skip and end IDs. The digital fader works on analog and digital inputs and outputs. Tapes can be time- and date-stamped using an internal clock.

SONY TCD-D10 PRO

This is the best-selling portable from Sony, and one of the only units on the market with a built-in speaker. Tapes can be time- and date-stamped using an internal clock. Although current machines do not have absolute time code, the unit is being upgraded to include it. It will not record at 44.1kHz using the analog inputs. Unfortunately, the device does not fully support AES/EBU specifications (factory update may be available) and has some clock jitter problems. As a result, the device will not work with SoundTools without a mod that voids it's warranty. It doesn't read program IDs from other machines, and indexes are not transferred during a digital dub.

TASCAM DA-30

This unit has 64× oversampling delta-sigma A/Ds and 8× oversampling 18-bit D/As. It has some digital clocking problems. Tapes recorded on almost any other machine won't have start IDs read correctly. If a tape has any type of copy flag, you will not be able to make copies using the consumer ports. With the AES/EBU ports this is not a problem. Margin display shows available headroom in decibels; it reads and writes absolute time code; indexes are not transferred during a digital dub; 32kHz recording is possible using digital inputs; has 15-pin parallel I/O port; and has +4/-10 signal on XLR-3 and RCA connectors.

TECHNICS DA-10

A best buy. Similar in many ways to the popular SV-3700, but uses lower grade converters. It has MASH noise shaping and 1-bit converters; analog I/O is -10 on RCA connectors; can stripe absolute time after the fact; and has built-in error-rate readout. Like the SV-3700, program IDs can be post recorded or erased. The DA-10 has coax and fiber digital I/Os; SCMS; but no AES/EBU. Digital fades only work on analog inputs.

If digital I/O problems have got you down, cheer up.

DIGITAL INTERFACING

If digital I/O problems have got you down, cheer up. There are a number of problem-solving interface boxes on the market that perform useful functions including copy prohibit bit stripping, SCMS elimination and format conversion. My favorite is a device manufactured by Digital Domain. (We like the name, too — Ed.) The FCN-1 format converter lists for $500 and will do conversions from consumer to professional formats including IEC-958, which is new type of consumer digital standard. Although IEC-958 is designed to be interchangeable with S/PDIF, it has a slightly different word clock, which can sometimes cause problems, according to Jesse Jacobson at The DAT Store in Santa Monica, CA. The FCN-1 will add or remove C-bits which allow the SCMS to keep track of how many copies of a tape have been made. It also adds and removes emphasis (early machines such as the Sony 2500 and the Sony DTC-1000 could only add emphasis). It also has a digital distribution amplifier, so you can go in from one machine and out to four machines simultaneously. It will also make sure that tapes have a sampling frequency flag and it claims to be compatible with future formats such as DCC and optical recording. Options include polarity inversion, digital "over" indication, and channel reversal in the digital domain. If your budget is a little tighter and you just need to strip out the copy prohibit flag, to record protected CDs, for example (professional, not-for-profit applications only, please!), another device called the Digital Inserter will do the job.■

Rick Schwartz and R•E•P would like to thank Jesse Jacobson of The DAT Store for his assistance in the preparation of this article.

Out on the wildest edge: audio design and post production for MTV's Liquid TV.

THE SOUNDS OF LIQUID TV

By Keith Hatschek

Liquid TV, airing in prime time this past summer on MTV, presented a unique, irreverent 30-minute blend of diverse animation segments by a variety of artists and writers. San Francisco-based Colossal Pictures originated the concept for the weekly series and Music Annex Audio Post Production, also of San Francisco, handled the audio for the shows. Described as 'dangerous, delicious and disposable,' Liquid TV parodies existing TV and comic book forms with no segment lasting more than three minutes.

Fifty-seven different segments comprised the first shows, so hundreds of audio elements on many different formats were delivered. In addition to the nine different Colossal Pictures directors creating segments, 13 other directors provided animated segments as well. All of these pieces had to be woven into a continuous tapestry of animation and sound by the Colossal/Music Annex team. According to Music Annex sound designer Jon Grier, "Having sat with pen in hand for months in silence, animators tend to have unique and well-developed ideas regarding sound. Each show was made up of 9 to 12 segments and various intros and bumpers that we were completing in a very compressed time schedule. Audio elements were flying in the door from directors in almost as many formats as you can imagine, but everyone retained their sense of humor ... after all, this is Liquid TV."

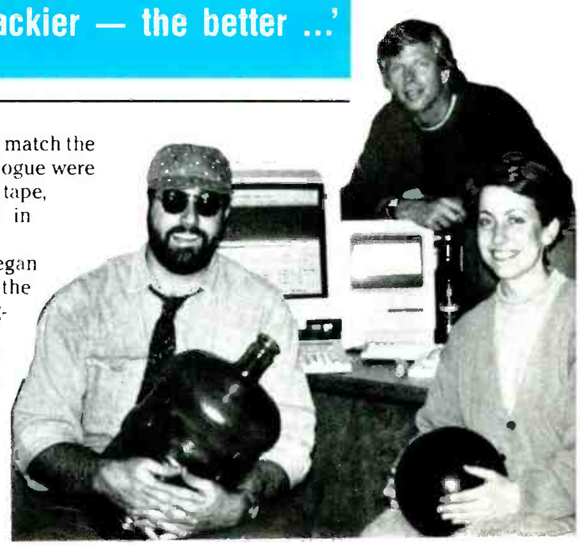
Keith Hatschek is vice president of marketing for Music Annex, Fremont, CA.

Initially, dialogue segments were recorded on 1/4-inch tape and transferred to mag stock for the directors. They then edited the original dialogue elements on film, and in a process called "Track Reading" assigned frame numbers (cel numbers) as a road map for animators to create pictures. Once animation was assembled, the dialogue was re-edited to better match the visuals. Final visuals and edited dialogue were then transferred from film to video tape, and off-line video editing began in earnest.

'Go for it ... the wackier — the better ...'

Shortly thereafter, off-line edits began appearing at Music Annex, and the track building process began. Segments varied from short subjects of 50 to 60 seconds, to longer pieces of two to three minutes. "Some source segments arrived with the audio elements on the tape and others just had dialogue. Producer Prudence Fenton and the individual directors gave us input as to what they heard for the individual segments. Then we rolled up our sleeves and got to the fun stuff," says Grier.

One of the principle areas requiring sound design was the innovative host segments that set the tone for the shows. "The host segments feature body parts floating in water with the host voice talking to the audience," says Grier. "In one host segment, a pair of lips is suspended in water talking ... these lips got to be known

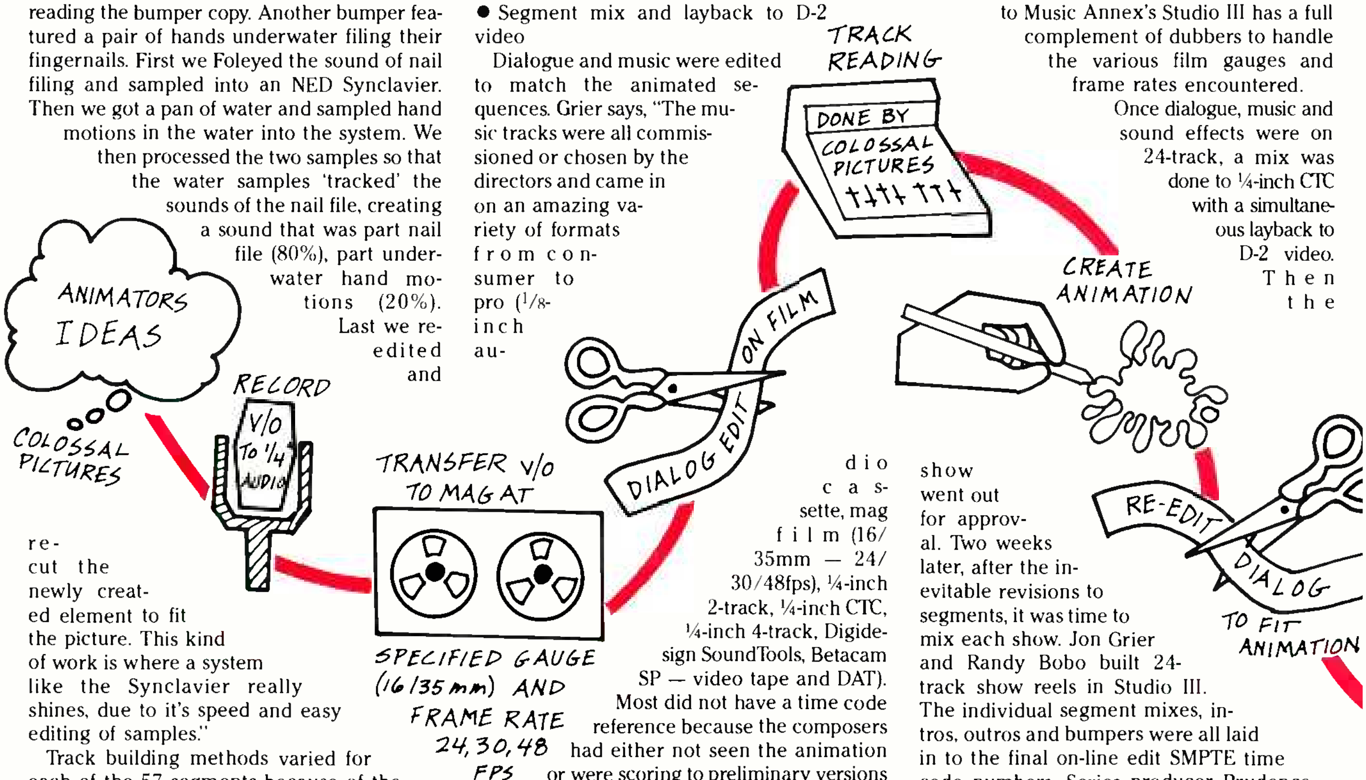


Liquid TV sound team Jon Grier, Will Harvey and Mary Ellen Perry in Music Annex, Studio III.

affectionately as 'Frankenstien Lips' and in keeping with their heritage, we recorded over 600 reads by nine different voices of the copy to match the lips ... things such as 'Stay tuned for Liquid TV.' Then we edited together a crazy quilt of different voices to match the lips reading the bumper copy. Another bumper featured a pair of hands underwater filing their fingernails. First we Foleyed the sound of nail filing and sampled into an NED Synclavier. Then we got a pan of water and sampled hand motions in the water into the system. We then processed the two samples so that the water samples 'tracked' the sounds of the nail file, creating a sound that was part nail file (80%), part underwater hand motions (20%). Last we re-edited and re-cut the newly created element to fit the picture. This kind of work is where a system like the Synclavier really shines, due to it's speed and easy editing of samples."

Track building methods varied for each of the 57 segments because of the differing styles each director employed. Music Annex staff performed the following tasks in this phase of the project.

- Dialogue recording / editing / dialogue replacement
- Sound effects editing/live Foley/electronic Foley
- Music editing
- Segment mix and layback to D-2 video



Dialogue and music were edited to match the animated sequences. Grier says, "The music tracks were all commissioned or chosen by the directors and came in on an amazing variety of formats from consumer to pro (1/8-inch audio cassette, mag film (16/35mm — 24/30/48fps), ¼-inch 2-track, ¼-inch CTC, ¼-inch 4-track, Digidesign SoundTools, Betacam SP — video tape and DAT). Most did not have a time code reference because the composers had either not seen the animation or were scoring to preliminary versions of the segment's animation. The director and I dumped the music into the Synclavier, and we edited the track to fit the segment. In some cases we used Synclavier's time compression/expansion capabilities to aid in re-syncing wild music tracks. Another piece of music did not end properly, so we found a toy piano sample, similar to the instrument used on the track, and just played out a new ending and matched levels and EQ for a fix. These types of changes would be almost impossible using tape. We would have had to call in the composer, and there just wasn't time."

High technology like the Synclavier wasn't the only thing to save the day in some instances. When creating a track for a segment titled "Footworks — Dog Flirting," where the picture showed the footprints of two dogs in a courtship ritual, unnamed Music Annex engineers rose to the occassion and provided Foley for the dogs breathing, panting, sniffing and even creating an effect fondly named "puppy piddle" using two pans of water to match the on-camera "action." Another segment entitled "Soap Opera," which details an unrequited love affair between a bar of soap and a liquid soap dispenser using stop-motion photography, required some quick Foley work as well.

"In one scene, the soap pump is getting aroused and we had to create a good sound for his pump, so we sampled the sound of a balloon being stretched and then played with the pitch in the Synclavier until it matched the on-camera action," says SFX editor Mary Ellen Perry. "Bringing the bars of soap to life was tremendous fun."

TRACKING THE CHANGES

Because of the varying amount of off-line editing that occurred to so-called "finished edits," often substantial changes occurred to what had been thought to be final dialogue tracks for a particular segment. Since off-line audio was for reference only, we found ourselves going back to original audio elements on mag or 1/4-inch and reconforming dialogue cues to new edits as they came in. Fortunately, the machine room adjacent to Music Annex's Studio III has a full complement of dubbers to handle the various film gauges and frame rates encountered.

Once dialogue, music and sound effects were on 24-track, a mix was done to ¼-inch CTC with a simultaneous layback to D-2 video. Then the show went out for approval. Two weeks later, after the inevitable revisions to segments, it was time to mix each show. Jon Grier and Randy Bobo built 24-track show reels in Studio III. The individual segment mixes, intros, outros and bumpers were all laid in to the final on-line edit SMPTE time code numbers. Series producer Prudence Fenton and executive producer Japhet Asher supervised the mixes done by Music Annex's Will Harvey. A few more changes were noted by Harvey in Studio IV and upon completing a segment, he'd alert Grier in the adjacent Studio III, who would make the changes and then re-mix the segment and drop it onto the show master.

"The layout of Studio III and Studio IV made these changes easy to perform and the producers could peek into both rooms alternately to monitor any changes, as well as keep tabs on the next show mix" says Harvey. "D-2 offers four channels of audio as well and we were able to lay the stereo show mix on tracks 1-2, a mono mix on track 3, and an M&E mono mix on track 4. With this layout the show was ready to go out for foreign language versions, with all elements on one piece of video tape."

As the mixing process headed into the home stretch, word came down that a U.K. version would be required. The air dates for the U.K. shows were very soon, so supervising engineer Randy Bobo dumped the completed American show mixes from the D-2 digital master into

Imagine a steamy soap opera performed by stop-motion animated bars of soap and a soap pump named Judd.

LIQUID TV SERIES CREDITS

MTV
Executive in charge of production — Abby Terkuhle
Creative supervisor — John Payson
Special thanks to — Judy McGrath

COLOSSAL PICTURES
Executive producers — Japhet Asher and Kit Laybourne
Producer — Prudence Fenton
Co-Producer — Nicole Grindle
Production coordinator — Amy Capen

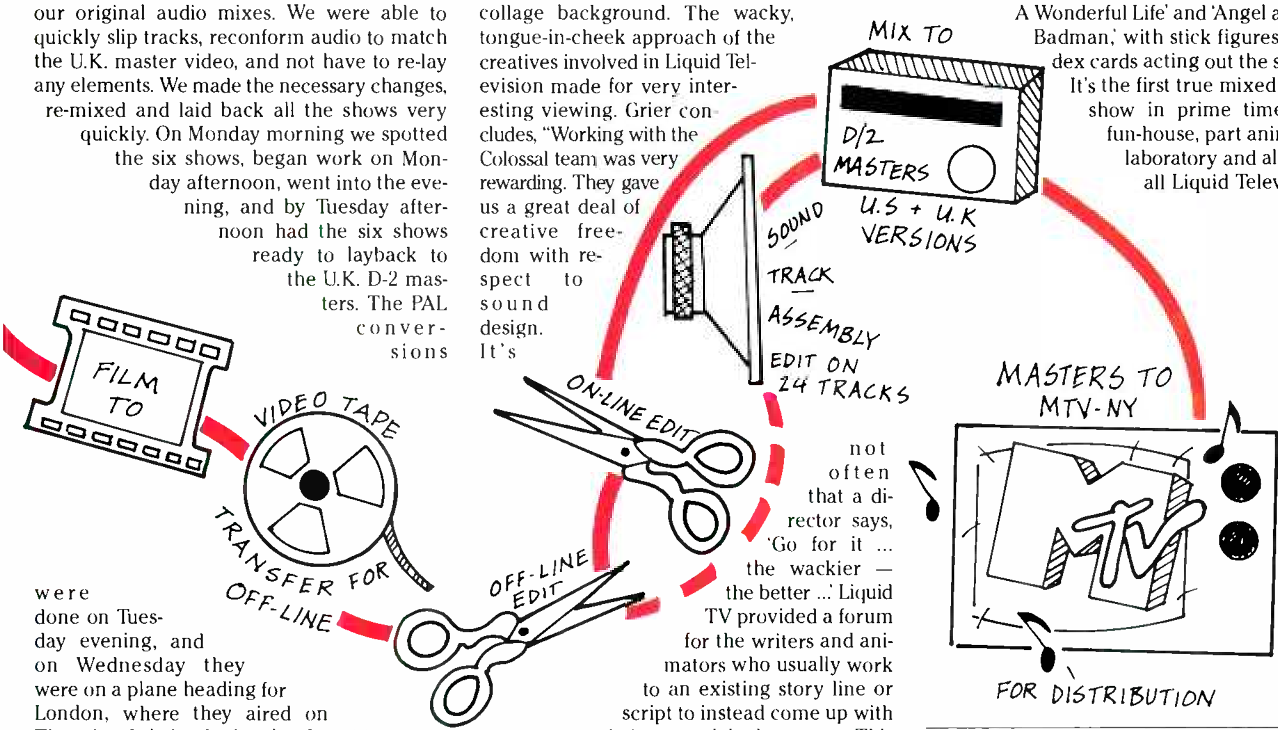
VIDEO POST · WESTERN IMAGES · SAN FRANCISCO
Senior editor — Pat Caballero
Editors — John Henkel and Alan Chimenti

MUSIC ANNEX AUDIO POST PRODUCTION
Supervising engineer —Randy Bobo
Sound design/segment mixes —Jon Grier
Show mixer — Will Harvey
SFX editor — Mary Ellen Perry
Transfer engineer — Linda Lew ■

Music Annex's Post Pro system located in Studio I for editing.

"The United Kingdom versions rearranged some of the segments and deleted a few portions, so that the new pictures did not match our original audio mixes. We were able to quickly slip tracks, reconform audio to match the U.K. master video, and not have to re-lay any elements. We made the necessary changes, re-mixed and laid back all the shows very quickly. On Monday morning we spotted the six shows, began work on Monday afternoon, went into the evening, and by Tuesday afternoon had the six shows ready to layback to the U.K. D-2 masters. The PAL conversions were done on Tuesday evening, and on Wednesday they were on a plane heading for London, where they aired on Thursday. It helped a lot that I could recall Harvey's mixes from Studio IV in Studio I," says Bobo. (Both rooms have Amek consoles with Master Mix automation.)

The wide variety of production techniques including cut-out, cel, stop-motion, stick figures, multi-plane and computer animation, blended live action against a collage background. The wacky, tongue-in-cheek approach of the creatives involved in Liquid Television made for very interesting viewing. Grier concludes, "Working with the Colossal team was very rewarding. They gave us a great deal of creative freedom with respect to sound design. It's not often that a director says, 'Go for it ... the wackier — the better ...' Liquid TV provided a forum for the writers and animators who usually work to an existing story line or script to instead come up with their own original concepts. This kind of project only comes along once in a great while. Imagine 'Soap Opera,' a steamy soap opera performed in a shower by stop-motion animated bars of soap and a soap pump named Judd, or 'Stick Figure Theater' which uses audio bytes from classic Hollywood movies such as 'It's A Wonderful Life' and 'Angel and the Badman,' with stick figures on index cards acting out the scenes. It's the first true mixed media show in prime time, part fun-house, part animation laboratory and all fun ... all Liquid Television." ■



For more information contact Keith Hatschek at Music Annex Duplication and Corporates Offices, 42650 Christy Street, Fremont, CA 94538; 415-226-0800.

Designing a flexible all-digital audio system for video post production.

By Paul D. Lehrman

VIDEO POST & TRANSFER'S DREAM

Far field at the first Dream system desk.

The digital audio revolution isn't just confined to CD production — the video world is going digital, too. "People are going to have to look a lot harder at digital audio, now that we have D-1 and D-2," said Neil Feldman, president of Video Post & Transfer, a leading post-production house in Dallas. The professional video formats he's referring to offer unprecedented audio quality, thanks to their four 48kHz digital audio tracks.

"In the world of 1-inch, the video machine acted as an audio filter," explained Feldman. "The quality could not exceed what that machine could do. When you replace 1-inch with a digital machine, that filter is gone. Now you can hear the limitations of the audio mixer and the electronics."

To maintain the audio quality the new formats demand, Feldman and his staff have been building a "Dream" — a Digital Real-Time Editing for Audio Mastering system — "to keep digital, digital." While pursuing that goal, they found they could solve another problem: keeping client costs down and minimizing production time.

According to Ron Evans, one of VP&T's audio engineers, "Clients wanted to mix in the post room. They might have only a couple of channels of audio, and they didn't want to go sweeten somewhere else. They wanted to be finished when they were finished."

Said Feldman, "We set out to keep things digital inexpensively, and in the course of putting things together, we realized we could do a lot more in the on-line bay."

The Dream system combines three technological advances of the past few years: hard-disk digital recording, the AES/EBU digital audio interface, and MIDI. It is built primarily around off-the-shelf components, which means its cost is considerably lower than dedicated systems, whose price tags inevitably reflect their manufacturers' R&D costs. It also means that the system is modular, and as new technologies and products become available, they can be incorporated at low cost without obsoleting the rest of the system.

At the heart of Dream are a Macintosh IIfx computer equipped with a 650Mbyte hard-disk drive with Digidesign's "SoundTools" system, and two Yamaha DMP7D digital audio mixers. The mixers are ganged together into a 16-input/4-output configuration. They use Yamaha's proprietary digital I/O format, so signals from the D-2 machines, SoundTools and a Studer 730 CD player, which are all in AES/EBU format, go through Yamaha digital format converters. The mixer outputs on their way back to videotape are also converted.

Analog audio decks — for instance Nagras and multitracks — as well as analog audio from 3/4-inch, Betacam/SP and 1-inch video decks go through the Yamaha A/D converters. "Whatever comes in," said Feldman, "we digitize and work with in the digital domain. Clients are getting sophisticated enough to know that if they are using digital mastering, they need digital all the way through — that's what they're paying for."

HARD-DISK AUDIO

Much of the audio work is done directly from D-2 to D-2, but when more tracks are needed, or when sound has to be moved around in time, SoundTools is used. SoundTools, Digidesign's 2-channel recording and editing system, consists of a hardware card that fits into one of the Mac's NuBus slots, an AES/EBU interface, and Sound Designer II software. It provides visual display of waveforms and a wide array of tools, such as mixing, crossfading, equalization, dynamic compression, time-slipping, time compression and expansion without pitch alteration. It records and plays directly from the Mac hard disk, with 650Mbytes providing more than one hour of stereo sound. It locks to SMPTE time code by first converting it to MIDI Time Code (MTC), a task handled at VP&T by an Opcode Studio 3 interface.

Evans, principal architect of the system, explained how it is used: "We can fly in sound effects from a CD in digital, stretch or shrink them, and move them around in time, with frame accuracy, then drop them over to D-2. Normally, the four D-2 channels are sync sound, voice-over, music and effects, but with the Mac we can also do stereo effects or music and keep it on line.

"We also use the Mac for music editing, and it does a great job with crossfades. We do dialogue replacement, and we can do both hard sync and 'squeeze' stuff to fit. Sometimes we do corrective editing on it, too. We had one track come in that was very distorted, and we were able to redraw the waveforms where the distortion occurred, and really clean it up."

Feldman added, "Repairing distorted audio is not a booming business, but it's something clients can use."

Off-line storage is handled by Digidesign's "DATa" program, that can archive files on any storage format which uses the AES/EBU protocol, such as the D-2 machines themselves. "We'll be using DAT in the future," says Evans.

Don Clark, one of the system operators said, "We're looking at magneto-optical disks, but they are still a little too expensive and too slow. But the more a client is into audio, the more they're willing to pay for the quality and flexibility of doing it all digital, and that includes the cost of storage."

Paul D. Lehrman, a long-time contributor to R•E•P, is a composer, author and consultant in the Boston area. He is on the faculty of the University of Massachusetts, Lowell, and is a member of the Executive Board of the MIDI Manufacturers Association.

MIXING AND MIDI CONTROL

The Yamaha mixers, which work entirely in the digital domain, do more than just combine the signals. Each channel has 3-band fully parametric EQ, and three internal effects buses, one of which can be patched externally. Moreover, they are entirely MIDI-controllable using ordinary note, program change, and continuous controller commands, with instant total reset and moving faders.

"Yamaha solved the automation problem, and at the same time prevented generation loss," Evans said. Even the EQ doesn't bring up any extra noise. We can now bounce sound back and forth without any signal degradation. Dubs are clones."

On the Macintosh computer is another Digidesign program, "Q-Sheet A/V," which is designed to handle MIDI-based studio automation, for controlling the mixers. The software, which also locks to MIDI time code, can work in real time, recording fader and other control moves, trimming and editing them. It also works in an edit-list mode in which individual events are entered and tied to frame numbers. Q-Sheet can also play back hard-disk files recorded with Sound Tools, so they can be shifted around in time as well from within the program. The software will even import a CMX edit list to make it easy to match audio events with visual ones.

The automation information and playback list is actually in the form of a MIDI sequence and is stored on the hard or floppy disk as a Macintosh file.

"It's really important to be able to store all the events and get to them later," said Evans. "People are always coming back to us with revision s for their projects, and we just call up the automation data from the disk."

The system also makes it easy to create and store multiple mixes for a job. Audio editor Pat Couch: "It's not an ordeal to listen to a mix two or three different ways, using different tracks on Q-Sheet. The clients have options as to what to use."

Although it's possible to control every parameter on the DMP7Ds with real-time MIDI messages, it's often easier to save a particular setup as a "scene" in the mixer's internal RAM or on RAM cartridges. These scenes can also be off-loaded into the computer using System Exclusive messages, and can then be stored as a file using Opcode's "Vision" sequencer and recalled quickly at any time.

ROOMS AND MONITORS

The creation of Dream has meant that VP&T has had to re-think its approach to room design. Edit suites now have to be acoustically constructed, because the final audio mix is being done in the same room as the video. "Now the room is the audio filter," said Feldman.

To assist in this, the Dream system includes a custom 4-channel monitor mixer designed by vice president of engineering Brad Walker. "It handles two different mixdowns at the same time," he said, "one for the main speakers, which the producer and video mixer hear, and one for near fields, for the audio mixer. It has 16 inputs to choose from, digital or analog, with independent front and rear mutes on each. Six

The creation of Dream has meant that VP&T has had to re-think its approach to room design.

inputs are active at one time. With the four outputs, we may be able to have it do surround sound in the future."

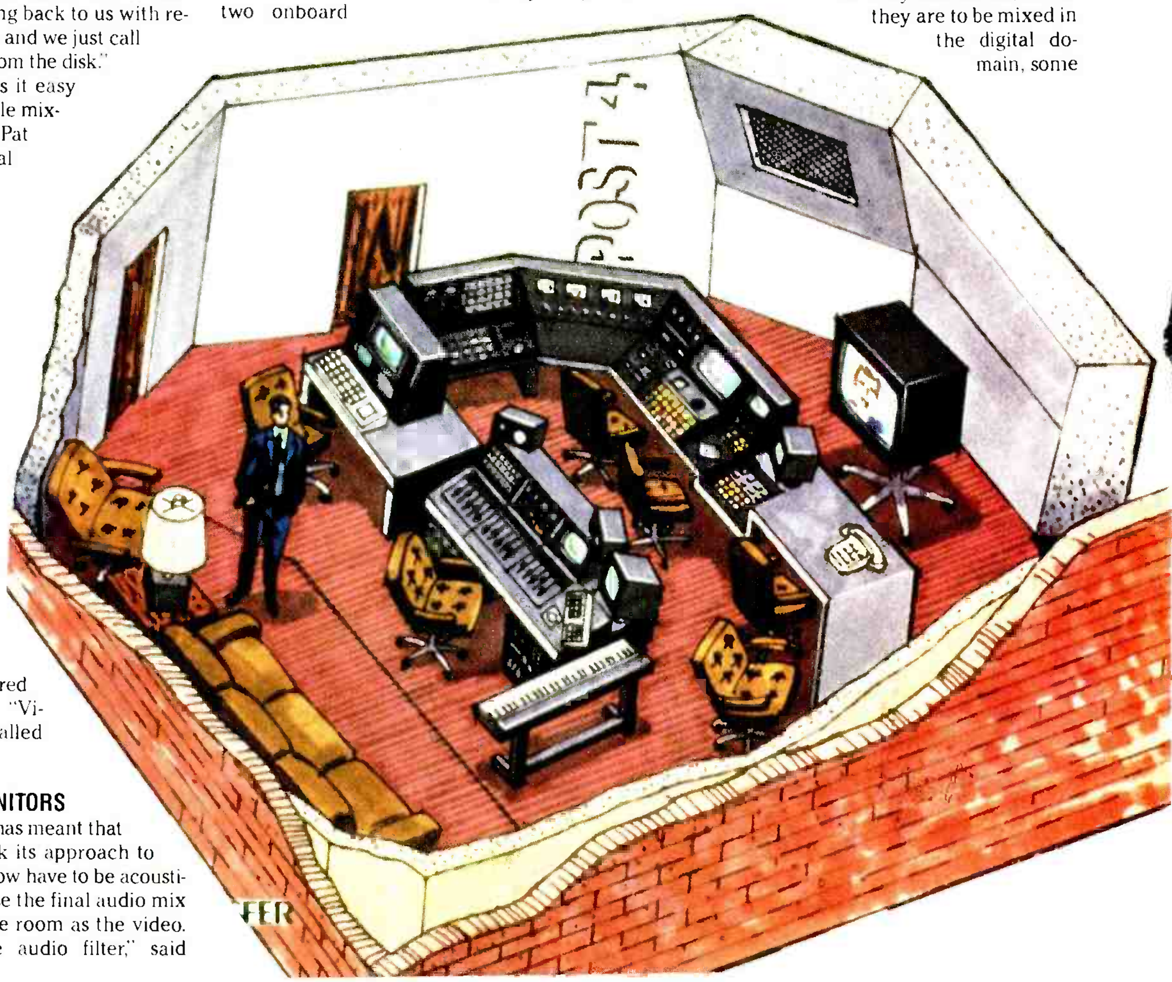
"We have to be able to use the four channels on D-2 however the client wants," said Feldman, "whether it's surround sound, multiple languages, or separation of the music and dialogue. The mixer has to be able to handle it."

There are two mono auxiliary buses for listening to any external processing that might be necessary (a Drawmer M500 compressor/limiter and Yamaha SPX-1000 effects processor are currently part of the system), and two onboard

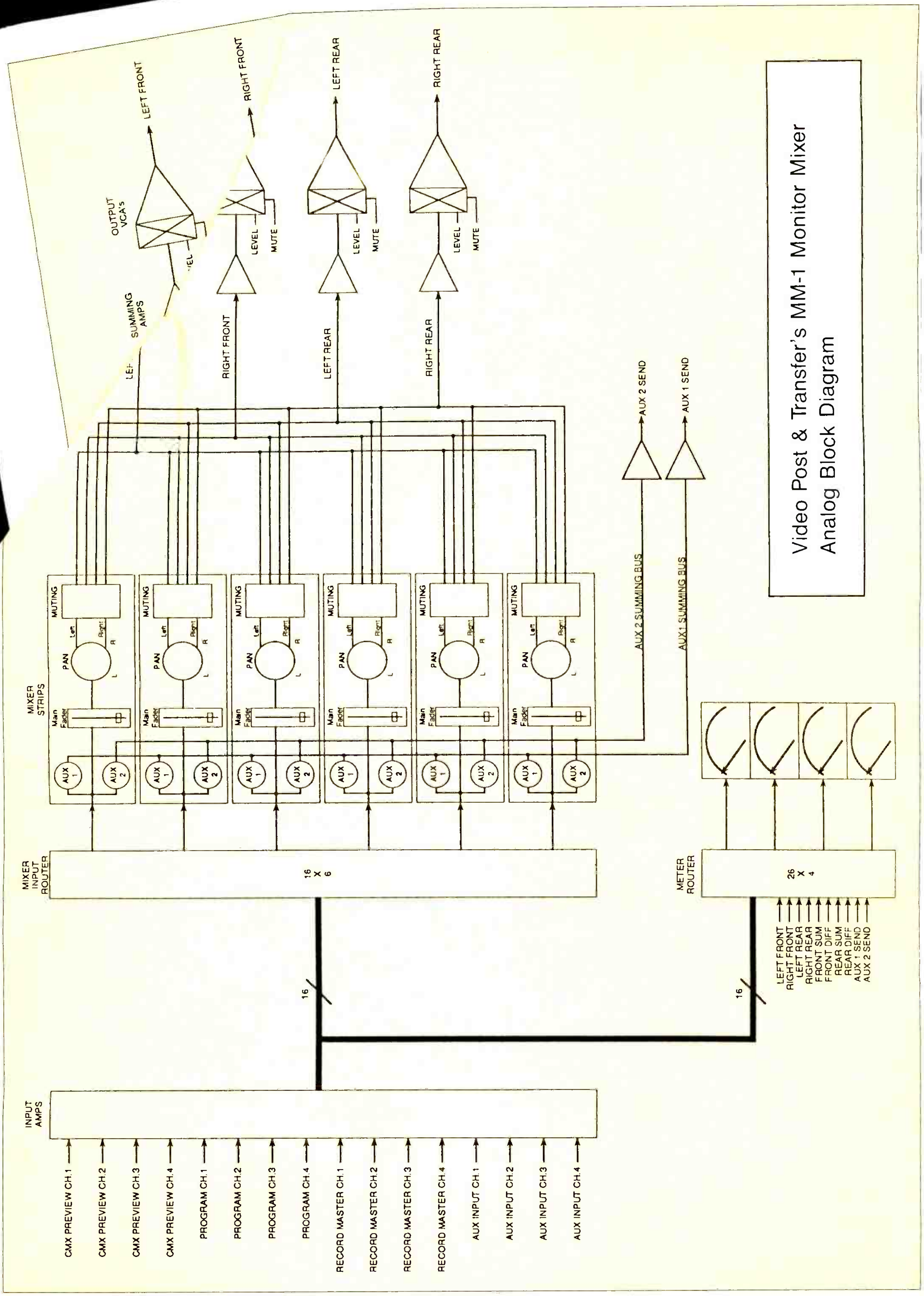
menories for signal routing. There's a digital
tonegenerator, with automatic speaker mut-
ing, ed in a tribute to the staff's favorite rock-
n-roll ovie, "This is Spinal Tap," monitor lev-
el con ls go to "11."
The xer can also be configured to follow
a video tcher so that it changes inputs when
the ur eck changes. A special mode is
available review editing in which the mixer
does not s when the pre-read starts, thus
maintaining r audio continuity. "The tech-
nology is no ew," said Walker, "but the
user interface
Metering on t m system is handled by
Dorrough digita m meters, which use
a large array of EDs to display pro-
gam and peak leve ously, referenced
both to analog and ns." In addition,
there is a Klark-Tek m analyzer.
Clark explained its use ital world,
the frequency spectrum is and we
need to be able to see all o can't
hear it. In a noisy environmen uite,
when people are talking, it's o
HF noise than to hear it."

SYNCHRONIZATION AND CONTRO

Another important task for the n. distribute word clock among the vari tal components in the studio. Synchroniz. signal sources in a digital audio studio i nearly as straightforward a task as syncing t machines, as VP&T has learned the hard wa The AES/EBU standard has no requirement fo. a separate word clock signal. This means that signals coming from two different sources may be unsynchronized, and if they are to be mixed in the digital domain, some



Future Dreamspace: The artist's rendition.



Video Post & Transfer's MM-1 Monitor Mixer Analog Block Diagram way must be found to force them into sync.

"We need to be able to take CD input at 44.1kHz," said Brad Walker, "and other sources at 48kHz, mix them together, and varispeed them. What we need is an AES/EBU receiver, a converter that will take any sample rate up to about 53kHz which is 48kHz with a 10% error and decode it, track it, and then output it to 44.1kHz or 48 kHz, locked to something that is independent of its own input: word clock, video sync, house sync or some other audio signal.

"Sony has a unit, the DFX-2400, that does it, but it takes up two rack spaces and costs over $10,000 per stereo input. The Yamaha converters we're using can take different signals that aren't perfectly in sync, but they must be at the exact same sample rate, and can't walk past each other."

VP&T is working with an independent company called GNP Research to develop a small, low-cost converter. Greg Basile, president of the company, explained further: "When you're using asynchronous AES/EBU generators, even if they're all using the same sample rate, you still get clicks and pops."

Walker added, "We would like to be able to lock everything to a master video sync signal, which is independent of the video tape recorder. Having the VTR as master is dangerous: If it goes into the wrong mode or powers down, it takes the whole room down."

The need for independent sync has caused some trouble for the facility when it comes to dealing with MIDI samplers. VP&T would like to use samplers as additional on-line audio sources. Q-Sheet can play back notes from a sampler, and even impose pitch change for added flexibility, and the post suites have Yamaha KX76 keyboards for "live" triggering of effects.

"We'd like to be able to fly them in from the keyboard," said Evans, "and then go back and trim them if we have to."

However, VP&T hasn't found a sampler that meets its requirements. The problem is that no sampler exists with digital outputs that accepts external word clock to drive those outputs. The principals are talking with several manufacturers to overcome this problem, and a solution may soon be at hand.

The digital audio revolution isn't just confined to CD production — the video world is going to digital, too.

THE FRONT END

One remarkable aspect of the Dream system that has emerged during its development is that the distinction between designer and user is breaking down. Thanks to the Macintosh and Hypercard, those on the front lines of audio editing at VP&T are increasingly involved in deciding how the system is going to work.

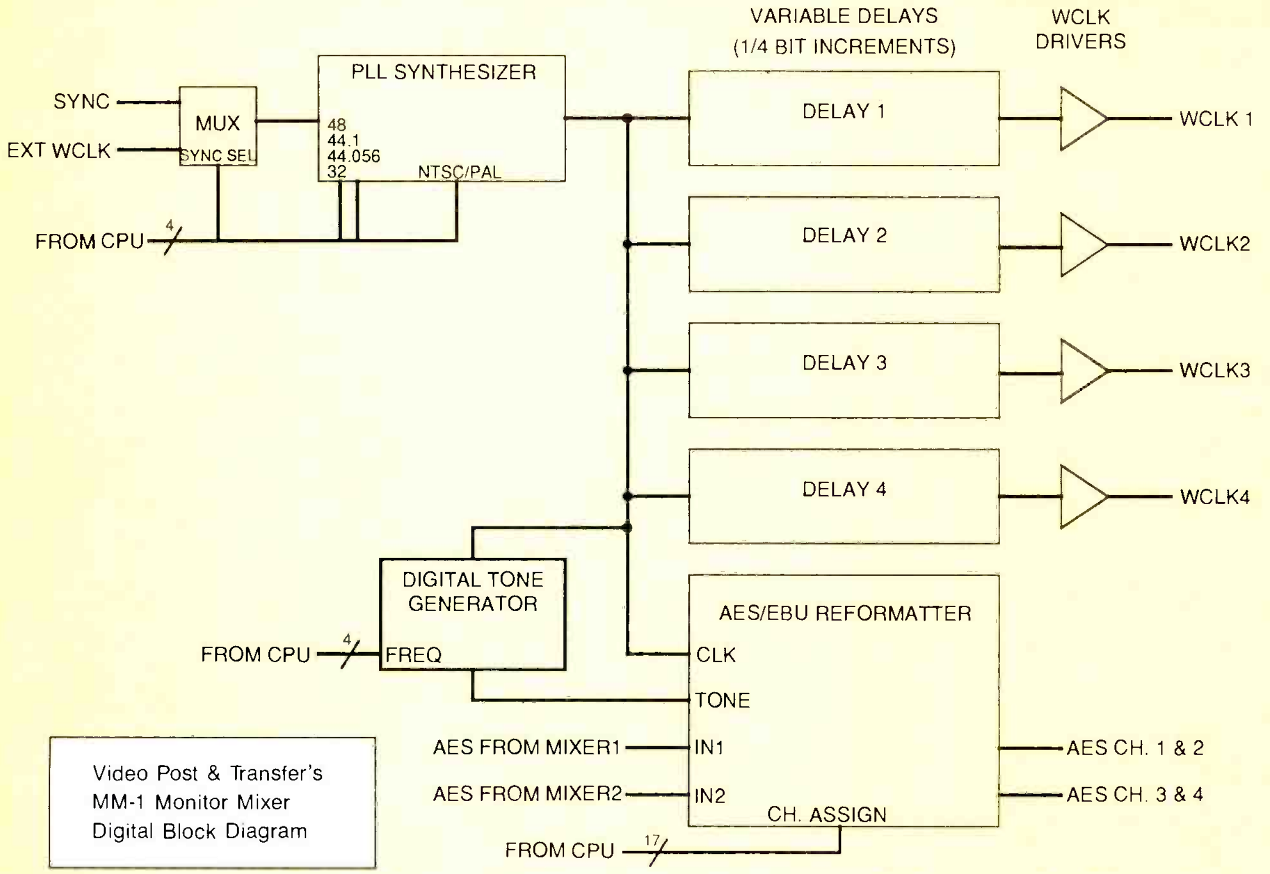
The front end of the system is actually a Hypercard stack, written by assistant editor Doug Wilson, which gives immediate access to the system's various functions, including the commercial Macintosh software (Q-Sheet, Vision, Sound Designer and Digidesign's "Softsynth" for synthesizer-like manipulation of recorded sounds), and a custom program for controlling the Studer CD player, using RS-422 serial protocols.

The CD driver program includes a database for on-line keyword searching of specific sound effects. The studio uses the Sound Ideas effects library, all of which is catalogued in the database. "You type in the keyword, and one or two sub-keywords, and the program finds it in the library and tells you what disc it's on," said Dan Clark. "You load the disc, and the computer cues it up."

Wilson added, "Our search function adds skip capability to the player. It can actually do it by itself, but Studer didn't put in a panel control for it."

The Hypercard stack is constantly being upgraded, to improve both its speed and flexibility. "Soon it will be controlling Nagras, D-2 and Betacam," said Wilson. "It will be able to handle tape shuttling, and you'll even be able to edit video directly from the Mac." It also has a time code calculator, a telephone directory, and a detailed take-out menu from a local eatery.

Other software is being developed to improve on Q-Sheet's capabilities. "Q-Sheet does 95% of what we want to do," said editor John Muller, who is writing a new program in Pascal, "but we sometimes need a better mousetrap. Q-Sheet can't trigger the CD player, for example,



Video Post & Transfer's
MM-1 Monitor Mixer
Digital Block Diagram

or any of the video machines. We also want a timeline display of mix parameters, with line segments for fades, and more automation. I want to be able to type 'Fade 30,' and have it automatically insert a one-second fade."

With all of this input, Dream is easy for a new user to get running on. When we visited the facility, Clark had been on staff only six weeks, and by his own admission "had never run a Mac before."

"The system is really easy to understand," he said. "It quickly stopped being a gadget and started being a tool. You can get your mind on mixing and assembling, rather than on which button to push."

Couch, who has been on staff about 18 months, sumed up: "The more anyone learns about it the more we challenge it, and figure out what it can and can't do. We can now do things we couldn't before."

THE FUTURE

Video Post & Transfer is going in two directions simultaneously with its Dream: The company has built a facility that includes three video suites containing Dream systems and one Dream-only room; and it is exploring the possibility of marketing turnkey systems to other facilities.

"It's about half the price of anything on the market now with its capabilities," said Feldman. "We're considering marketing a second-generation system for about $90,000, total, which includes the recording system and the mixer. Compared to what's out there, it's ridiculously low.

"Before, if you wanted to bring audio post into the video suite, you'd probably have to put half a million dollars of audio equipment in — when you total up the synchronizers and other kludges — but now it can be done for less than $100,000. And that means you don't have to raise the price of the room. That is what we're doing at our new facility, and what we hope others will want to do."

With all of this input, Dream is easy for a new user to get running on.

The systems at the new facility will eventually have 24 or possibly 32 inputs. "Ganging more Yamahas is always possible," said Feldman.

The facility, about a mile from Dallas' Love Field Airport (where the company was formerly located), opened in June. The architectural renderings for the new plant were done in-house on the Macintosh, using Claris CAD software and a Houston Instruments color plotter. (Not surprisingly, the system has seen additional use in designing the front panel of the monitor mixer.)

Hand in hand with the low initial cost is the system's modularity, which means it will be inexpensive to keep up to date. "It is easy to change with the times," Evans said. "The beauty of it is that instead of being locked into one big box from a company, if Yamaha or another manufacturer comes up with a new mixer or converter, we can replace just those components."

Feldman said, "With big dedicated systems, you have to wait for the manufacturer to make the feature, and you're locked into them. The advantage of the Dream is that this is our first pass. When new equipment comes out, we will change."

Evans concluded, "If there's a new Mac program, a better CD player, a cheaper storage device, or better processors, we can add them just by plugging them in. It means I can continuously improve the quality of the system without any significant increase in the overall cost."

Video Post & Transfer may have hit on a unique solution that solves many problems at one time, and will go far toward realizing the potential for superb audio quality to match the new high-quality video formats. If the company is successful, its Dream will become reality for many facilities and clients. ■

Tour Business Slowdown Prompts Diversification

By David Scheirman

The previous concert touring season has been one of the "softest" in the past decade, and many live sound professionals and concert touring companies have looked to other related business endeavors to ensure a steady cashflow. With the reluctance of many concert acts and promoters to tour at the beginning of last year due to the Persian Gulf war, the industry never quite got back on its feet before the summer season was upon us ... and the year stayed soft.

1991 saw very few "blockbuster" shows such as those that drew record crowds during 1989-90 (including major tours by Paul McCartney, Phil Collins, Steve Winwood, The Who, The Rolling Stones, Madonna and such). In '91, many of the outdoor amphitheaters had unusually quiet seasons, and there were literally no nationwide stadium tours booked with the exception of some late-summer dates by bad-boy rockers Guns 'n' Roses, a collection of major outdoor shows by perennial favorite Jimmy Buffett, and a successful tour by the Grateful Dead.

"We decided to go ahead with our plans for stadium dates, despite the economic slowdown," said Cameron Sears, tour manager for the Grateful Dead. "It turned out to be a killer year for us. No one else was out doing the major shows, and we ended up with record-breaking crowds in many markets." Indeed, the Dead were consistently rated as the #1 concert act draw throughout 1991 by *Amusement Business* ... and many of those shows were stadium dates.

Many other major concert acts touring in arenas and other large venues were not so fortunate, and many saw less-than-capacity crowds. Even well-established acts such as Paul Simon, Diana Ross, The Judds, The Moody Blues, Tom Petty & the Heartbreakers, Aretha Franklin and Manhattan Transfer reported audience draws of 55-70% capacity for shows in some regions of the country.

The touring business slowdown saw the rise of an innovative new trend: "package tours." Creative promoters put together "packages" of different bands that would all travel together and perform on the same stage ... sometimes as few as three or four acts, sometimes as many as eight or nine. As was the case of the Lollapalooza Tour which featured artists like Souxsie & The Banshees, Living Colour, Jane's Addiction, and Ice-T. One of the unexpected tour successes of the season, the Lollapalooza Tour drew large crowds to major venues, giving exposure to a variety of rock acts that would not normally have been playing to such large houses. Other hastily-assembled tour packages, some more successful than others, featured oldies acts, heavy-metal rock groups and jazz artists.

When the touring business softens up as it did in '91, what happens to companies that are used to an active concert business? Many such companies are finding that it is very important to branch out into other profit-generating directions in order to keep gear and personnel working. High on the list of alternative endeavors: permanent installations, long-term leases or sales of gear to civic venues and specialized product manufacturing for retail distribution.

A close look at some major touring companies such as Clair Bros. Audio and Maryland Sound Industries shows the year-round operation of dedicated permanent systems sales and installation divisions. The design, sales and installation of turnkey sound systems for venues such as outdoor amphitheaters, city auditoriums, theme parks, major churches and government facilities helps to keep road crews active during slower seasons, and leads to new opportunities for employees looking to broaden their skills in the area of system design, crew supervision, fabrication and such.

"Our installation department is only in its third year, but business is booming," said Gene Clair of Clair Bros. Audio. "There's always something cooking. We have staff that works on nothing but that, but our road guys can jump right in when they are not out on tour, and provide project management and expert training for our installation clients."

Some concert sound companies find that the high-tech field of doing special events for industrial clients (new product introductions, national sales meetings and conventions, and corporate retreats) offers a new market for both their specialized audio skills and their rental systems inventories. While this field (a.k.a. "corporate theatre") is often controlled by long-established companies in New York, Chicago or Los Angeles, there are significant opportunities to break into this field for regional companies and their personnel. Oftentimes, an audio-visual consulting company or event producer seeks sound system resources in different parts of the country. For instance with a Broadway show, a production "design" is generated for the corporation's events, and then locally available crews and sound systems are sought for use in convention-area locales such as Phoenix, Las Vegas, Orlando, Honolulu, Dallas and other resort cities.

"There are very high quality expectations in this field," said Bruce Cameron, an independent sound system designer/operator based in New York. "The equipment requirements are getting more complex. I just completed a series of shows for IBM and we had top-caliber rock entertainment talent such as Huey Lewis & The News; week-long shows for the company's sales force were staged in Florida and California. These events require a blending of resources from the audio-visual and the concert audio industries."

Some companies find that creating a specialty "niche" for their services can keep them in demand ... and in the black. For some, such as Compact Monitor Systems in Los Angeles, this may mean the design, construction and rental of stage monitor systems that take up as little truck space as possible. For others, such as Hi-Tech Audio in San Francisco and Mercury Sound Leasing in New York, mixing console rentals may be the key to notoriety. Still others, including ATM Audio in Carson, CA, are actively pursuing spin-off manufacturing arms. (ATM is the creator and manufacturer of ATM Fly-Ware, a line of rigging products for the modular speaker enclosures now supplied by a variety of speaker manfuacturers.)

Perhaps nothing offers as great an opportunity for regional sound companies as the growth of community-sponsored festivals, fairs and concert series. With many city governments and regional arts councils seeking ways to boost local culture and pride, and to create new draws for tourist trade, a wide variety of special events are becoming known around the country.

Events such as the Telluride Bluegrass Festival, New Orleans Jazz & Heritage Festival, Chicago Blues Festival and many others are drawing larger and larger crowds each year. This means that there is a requirement for more sophisticated sound reinforcement systems and the qualified crews to operate them. When these regularly scheduled events are combined with the rise in popularity of benefit-type rock concert events to aid charity and non-profit causes, a significant regional market can be seen in many cities that has little or nothing to do with concert "touring"... and yet, the sound system requirements are virtually the same.

The 1992 season should see a change for the positive, with many artists returning to the road after a few seasons of down-time (John Mellencamp, U-2 and a re-formed Journey to name but a few). And, there are new "mega-tours" reportedly in the works. (Genesis will tour to support their new album, maybe Madonna again, perhaps Bruce Springsteen, and there are even whispers [and denials] of Michael Jackson's show plans for '92.) However, the mega-tours belong to the mega-companies, and often such rumors don't turn into reality.

With as many as a dozen or more arena-sized concert sound systems to keep busy, the major companies are keeping a watchful eye on the entertainment industry's plans for the near future. It's significant to note that many veteran concert promoters are picking and choosing their booking commitments very, very carefully. For smaller sound companies, regional touring companies, hard-working tour individuals and everyone else in the business, wisely chosen diversification activities and a hope for the return of a booming economy are on the schedule for 1992. ■

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

Sound reinforcement and event production for an industrial product release — with an English spin.

INDUSTRIAL

The $1.8 million GM/Vauxhall product launch used an extensive audio-visual show to introduce the updated Astra to thousands of delegates from 600 dealers.

ROLLOUT

England's GM/Vauxhall automobile manufacturers are updating their small-car Astra series to take them deeper into the European fleet car market. To take the news to their dealers, they recently organized a series of product launches at the new International Conference Centre in Birmingham, England, inviting a total of 7,000 delegates over a period of six days.

Organizing the sound on behalf of production company Spectrum Communications Ltd, Delta Sound Inc. designed a sound system around the new Midas XL3 desk, taking full advantage of its VCA control and multiple output facilities.

The launch show used a combination of A/V and 35mm film interspersed with presenters from Vauxhall's management team — all leading up to the final product reveal.

The sound was a combination of multitrack for the A/V and product rollout sequences, stereo for the film sequence, walk-on/walk-off background music and radio mics for the presenters.

Tim Frost is an international free-lance journalist specializing in the audio, broadcasting and computer fields.

In total, the system demanded 13 independent loudspeaker zone outputs. It was this element of the specification that tempted Delta's Managing Director, Paul Keating, to move away from his traditional consoles and try the Midas.

"On productions of this size we normally require a higher level of outputs compared to the number of inputs. With the 13 feeds to speaker zones we needed a console with around 16 output buses, and the Midas having 22 separate controllable outputs gave us even more control. The alternative of busing two smaller boards together would have been impractical from an operational viewpoint and without the VCA control, we would have been manually trying pull down 16 outputs in a split second.

TAPE TRACKS

The main recorded sound source for the show system came from a Fostex G16 16-track deck, which contained all of the music, effects, film sound and voice-over elements, each recorded on separate channels premixed for the separate loudspeaker chains. With time code coming off tape, the G16 became the master machine for the event, with the code controlling the A/V slide system, film, lighting and some of the mechanics.

For immediate safety backup, a Tascam 34B with a stereo mix of the soundtrack and additional time code track was run in parallel. If the G16 failed in the middle of a section, then they could immediately fade over to the 34B until the next presenter support section. At that point they would connect in a complete duplicate standby G16/34B system and continue feeding the show from a multitrack source.

This double level of backup extended to other areas. Each of Vauxhall's management presenters wore two complete Micron radio mic systems, and there were also cabled mics to hand off should both of the radios fail for some reason. With $1.8 million invested in the launch, and 600 dealers waiting to be impressed, no-one wanted to risk any foul-up mid-presentation.

The electronic music used throughout was commissioned specially for the show, and the composer had supplied various stereo submixes and sound effects sections on DAT tapes. These came without time code and were laid off onto an 8-track DAR Soundstation and then synced together to create the full soundtrack.

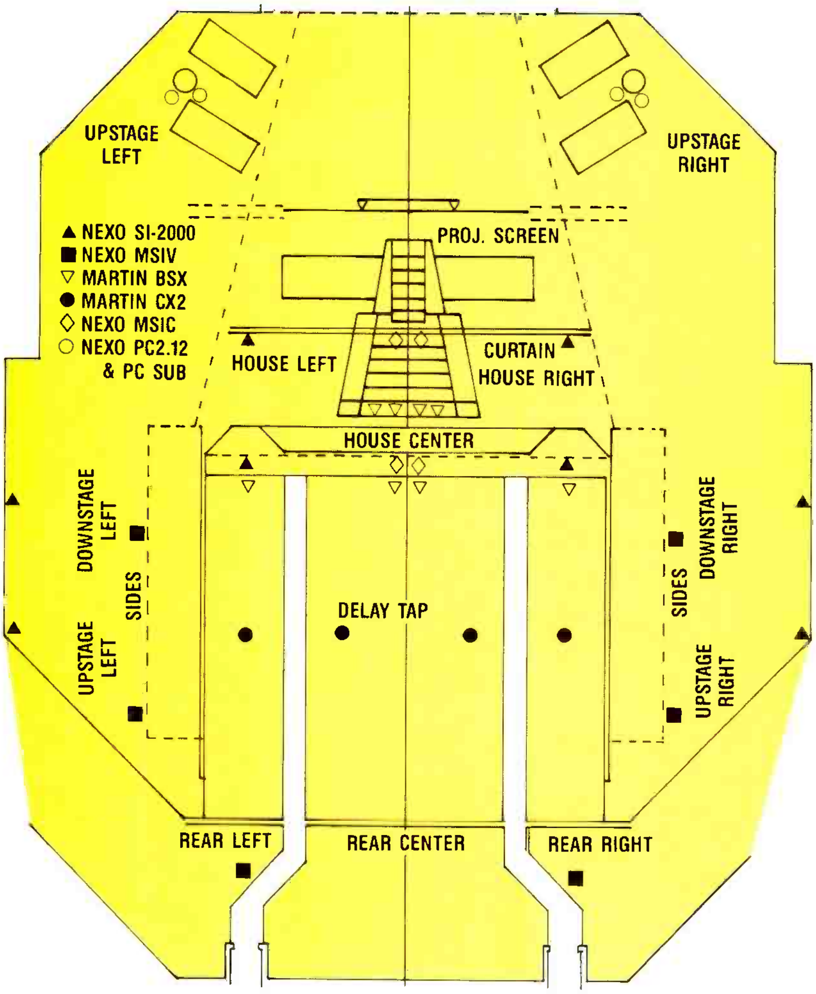


Figure 1. To increase the impact of the presentation, the main seating area was surrounded by false walls, which lifted away to reveal the cars placed around and behind the stage area. The loss of the hall's depth placed the presenters only 10 feet from the front row of seats.

All of the signal sources were routed first to the main racks and then distributed from there to the others via multicores.

The speakers were flown at 22 motor hoist positions around the hall's 43 feet high ceiling, which comprises a network of steel grids with access for the riggers.

The control room housed a 40-channel console with 22 separate controllable outputs, along with a ½-inch 16-track, which contained all of the music, effects, film sound and voice-over elements, each recorded on separate channels and pre-mixed for each of the loudspeaker chains.

During the four days of remixing at the London-based Soundtracks studio, Delta used a speaker layout that mimicked the final system as much as it was possible in the studio environment. Dedicating one tape track per speaker channel, they created a final mix where the bulk of the fades and effects were already placed on tape. This meant that very little mixing and additional effects work needed to be done during the show itself, simplifying the operator's job considerably.

The live effects were primarily small amounts of stereo echo and reverb added to the side and rear effects channels to give more 'depth' to one or two of the sequences, while firmly maintaining the focus on the picture rather than distracting delegates with noises coming from above or behind them. Keating had considered using MIDI control on the Yamaha outboard gear to automate this function, but felt that MIDI presented some potential software problems when set-up time was at a premium. So the occasional effects were added manually using the effects return channels routed to the relevant speaker positions.

The 13 main and effects tracks were recorded from the 8-track Soundstation, back onto tape in two passes, using the time code track to ensure that they stayed precisely in sync.

The G16 tape contained all of the recorded sound for the show, including the soundtrack for the 35mm film, with each channel being fed to an input on the Midas.

In addition to the 16 tape inputs, the XL-3 handled 6 channels of microphones, 9 channels dedicated to back up systems, and assorted returns and stereo inputs, using up all but a couple of the XL3's 40 channels.

CONTROLLING FACTORS

On the console, mics, FX, music, radios and backups were each allocated to an individual VCA. The master VCA pair was set up to control the overall output of the desk and used for master fading functions.

The ICC's Hall Three, the venue for the launch, is some 60m deep and 40m wide, and to increase the impact of the product rollout, the main seating area was surrounded by false walls, which with the stage surrounds, lifted away revealing the full range of cars placed around and behind the stage area. This meant that a lot of the hall's depth was lost for the main presentation, with the seating area becoming 23m wide and only 20m deep, and resulting in presenters being only 3m away from the front row of seats.

With virtually no depth to play with, the biggest problem facing Delta was to design a sound system that offered adequate coverage and level without running into feedback problems. This was further complicated by the fact that the live presentation elements were given by Vauxhall executives not speaking from a fixed position but moving around the stage area.

Normally for a presentation, Delta would install a split system — a screen position for music, and a separate vocal system for the radio mics. But without the depth of stage and seating area, they had to use the main A/V system for the radio mics as well. With loudspeakers placed 5 meters upstage of the main presenting area, level, especially when using the omnidirectional radio mic capsules, became Keating's most critical concern.

"When you are dealing with non-professional presenters you need to have in hand as much headroom in the system as possible, and the headroom situation was certainly not helped by the upstage presenter positioning. To resolve this problem we controlled the coverage by making sure that no one loudspeaker system was used to cover more than a 10m depth of seating area."

SPEAKER SYSTEMS

The twin main speaker systems used Nexo units, each with a pair of SI2000s for stage left/right and a pair of the smaller MSICs for stage center. The systems covering the rear half of the seating were mounted high at the front of the stage, but the system covering the front half was the one set 5m further back, deep within the set and well behind the presenters.

To assist the clarity of the speech elements, Delta put four additional Martin Audio CX units half way down the auditorium, and these were fed from the mic channels and the voice-over tracks of the A/V and film sections.

These, like the main speakers, were fed via delay lines, a mix of Klark-Teknik, Yamaha and BSS units. Delays of between 16mS and 42mS were applied to focus the speech elements to a main reference point downstage center. The only advantage of having the loudspeaker systems behind the presenters was that Delta didn't have to bother with any form of foldback, as the presenters could hear perfectly well from the main system hanging behind them.

The rest of the loudspeaker layout was primarily concerned with atmospherics and effects, with additional SI 2000s on the left and right exhibition hall walls. Nexo MSIVs were hung at the rear and along the sides of the false auditorium walls, and a total of eight Martin Audio BSX sub-bass units placed under the stage and strapped under the seating.

Every speaker channel had its own 1/3-octave equalizer tuned using a mix of ears and a dbx RTA-1 analyzer. On the main speakers, the EQs were set relatively flat, with no boost, and cuts of up to 3dB at the most. Firing through the heavy cloth of the false walls, the side speakers needed a certain amount of HF

TECHNICAL DETAILS

Tape Track Allocations for Fostex G16
1 Front left
2 Front right
3 Front center
4 Left side
5 Right side
6 Rear left
7 Rear center
8 Rear right
9 Exhibition left
10 Exhibition right
11 Sub FX/Drones
12 Voice-over cine
13 Mixed cine
14 Blank
15 Backup SMPTE time code
16 Master SMPTE time code

EQUIPMENT LIST
Mixers
1 Midas 40 channel XL3
1 Soundcraft 8000 16/8/8

Tape Machines
2 Fostex G16 1/2-inch 16-track
2 Tascam 34B 1/4-inch 4-track
1 Tascam 112R Cassette
1 Tascam 401 CD
1 Sony FS55 DAT

Processing
3 C Audio 312R stereo graphic
3 Klark-Teknik DN300 graphic eq
4 Yamaha 2031 EQ
2 Klark-Teknik DN410 parametric EQ
1 BQ3 parametric EQ
1 BSS TCS 804 Delay
1 Yamaha YDD2600 Delay
2 Klark-Teknik DN716
1 SCV Pro Source Director
1 Yamaha SPX50D
1 Yamaha SPX 90 II
1 Yamaha SPX900
4 Dolby SR noise reduction cards
2 dbx 150X noise reduction cards
1 AKG TDU 7000 Delay
1 dbx RTA analyser

Microphones
6 Micron diversity radio mics
6 Sennheiser MKE2 capsules
2 Sennheiser MKH40
2 Beyer M400

Speakers
8 Nexo SI2000
4 Nexo MSIC
7 Nexo MSIV
4 Nexo PC212
4 Nexo PC sub
6 Martin CX2
8 Martin BSX
4 Fostex 6301 self powered

Amps/Speaker processors
12 Yamaha PC2602M
6 C Audio SR808
3 C Audio SR606
2 C Audio SR404
4 C Audio TR850
11 Nexo processors
2 Martin EX2 processors

Multicores
1 100m 9 pair Belden
4 100m 19 pair Belden
6 50m 6 core speaker
4 Nexo SI2000 racks

lift, but even this was limited as these channels carried effects only, and intelligibility and accurate tonal balance was not a major issue.

The RTA-1 was one of Paul Keating's most-used tools, being employed for a lot more than simple EQ measurements.

"We use the RTA to analyze the response of each of the speaker clusters from a central point. The RRC room response facility allows us to compare the input to the speaker against the output in the room, which is helpful. But we also use the dbx system to check mic capsules for consistency and for tape machine line-up."

Hard disk system reliability is now good enough for live work and the cost is coming down.

Keating was also very happy with the design, access and facilities of the new hall. Set-up was over a period of three days, with the first day allocated to system installation, the second day for control and the third for system testing and alignment.

In all, 22 motor hoist positions were used to fly the speakers. Although 13m high, the hall's ceiling has been designed as a network of steel grids with access for the riggers to get almost anywhere in the roof area. Unlike some venues with a restricted number of load-bearing points, this grid allowed Delta to fly speakers wherever they needed.

The only omission in the hall's facilities is a service lift to take hardware to the balcony operations position. The Midas got there by 'Mandraulics,' Delta's term for the crew members taking a corner each and manhandling the console up into position.

Cable trunking is also well catered for at the ICC. The main racks driving the Nexos were at the rear of the hall, 20m to the side of the mixing position, with additional Yamaha-based amp racks back-stage and by the projection gantry.

The main racks contained the Nexo electronic speaker system controllers running the S12000s bi-amped from C-Audio SR series amplifiers, and all of the signal sources were routed first to these racks and then distributed from there to the others via satellite multicores.

Instead of using a separate mixer output channel for the sub-bass, the main outputs were routed first to an SCV Source Director which passed the signal unaltered onto the main system and central clusters, whilst generating a separate low pass filtered mono output for the Martin BSXs.

SHOWTIME

On performance day, everything went smoothly. Back-up was required only once when the synchronizer controlling the 35mm projector failed, unlocking it from soundtrack coming from the G16. For this eventuality Delta had an audio follower in the projection gantry with a duplicate of the soundtrack recorded using Dolby SR, so when they switched over to it, there was no quality loss.

Having used the DAR Soundstation for preparing the soundtracks Paul Keating is now developing the idea of using a 16-track hard disk system for playback on-site at a similar-sized event next year. Using a Soundstation for preparation and on-site would mean that different work practices can be brought into play. The show could be constructed from the outset so that sections would be easily updated at the last minute, and balances, that are going to be venue dependent, can be quickly tuned on-site.

Hard disk system reliability is now good enough for live work and the cost is coming down. But the flexibility to cut and paste, and adjust fade times and track allocation on-site, Paul believes is going to have to be explained to the producers who will initially just see it costing ten times more than a tape machine.

"Within the conference industry you are up against time. You don't have theater schedules with a week of previews; you have a couple of technicals, a couple of rehearsals and then, in this case, straight into a £1 million ($1.8 million) show which makes or breaks in 45 minutes.

As well as the sound quality advantage, a hard disk system will allow the producer flexibility around a carefully pre-programmed production. They will have much more power to fine tune their productions within the limited time they have on-site." ■

HANDS ON: PEAVEY DPM 3 SE

By Mike Oxlong

Debate continues over the viability of music "workstation" platforms for professional and private use. Key to this debate is the price vs. performance feature continuum. Second-generation music workstations are in use, and third-generation units are close.

At issue is the comparative hassle of connecting discrete components to assemble a personal studio. Multiple keyboard controllers, FM or alternative sound generating modules, sampling keyboards and/or modules, a drum machine, outboard effects, plus a MIDI mixing console, and an external sequencer or PC computer are required. Add interface software and hardware, and this larger-than-a-roadcase proposition gets pricy. Technical expertise is mandatory.

As alternatives to integrated MIDI studios, workstations bundle components and functions into a stand-alone professional platform. Less cash (often under $5K) is needed. Learning curves (one box, one manual, one nation under God) are shorter. And rather than honing your retail purchasing/deal-making skills, more time is spent doing real work.

THE DPM 3 KEYBOARD FAMILY

While getting past the Peavey M.I. image might be hard for some, most professionals are now aware of the user-driven innovation springing from Peavey. Cornerstone to Peavey's first all-in-one workstation is the DPM 3 Composition Center. It looks like any other live performance MIDI synthesizer. The DPM's software-based architecture engages three powerful Motorola 56000 series chips and is aimed at recording applications and artist-oriented MIDI production studios.

Peavey has merged the sound production capabilities of several analog, digital and sampling sound modules to an onboard 9-track, 20,000-note sequencer. Also included is 24-bit processing from four internally controllable "effects racks." As with many MIDI mixers on the market, track levels can be controlled via the sequencer's sends and automated amplitude controls.

Sampling rates of the DPM 3 deluxe version (known as the DPM 3 SE) top out at 48kHz/16 bits. DPM 3 SE sample input is accessed via its 1-rack-high companion: the DPM SX. A professional front-end A/D converter, the DPM SX features both SCSI and MIDI Sample Dump Standard (SDS) interfaces. These protocols allow SX samples to be shared with third-party sampling technology.

Once a sample has been recorded by the SX module and routed by MIDI cables to the 3 SE, sounds can be edited in much the same manner as "resident" DPM 3 SE ROM sounds. Digital effects processing and automated mixing of sound files (once they're recorded to sequence) complete the package.

The DPM 3's comprehensive manual, written by MIDI guru Craig Anderton, details the potential of these machines. The front-panel control surface is separated into five primary sections: system controls, voice editing, sequencing, data control and performance control.

SYSTEM CONTROLS

In the upper left-hand chassis corner, six push-buttons access 100 on-board user programmable (RAM) sound files. Sounds are matrixed into 10 banks of 10 files each. Individual sound files can be constructed of on-board factory ROM waveforms, sound files from sample libraries, including Prosonus (stored in RAM), or user samples loaded from the SX (which are also stored in 3 SE RAM).

Whereas most commonly used factory and third-party sound files offer multiple sources, the DPM's production value is enhanced by access to user samples.

The DPM is defined by the concept of sound file. Single sound files may consist of a variety of basic sounds. 'Basics' include two sound-generating oscillators that draw from ROM and/or RAM sound files (i.e. factory vs. user-supplied samples/libraries).

Each sound file can then be "combined" with up to three other sound files (each with another two oscillators), taking advantage of up to eight separate oscillators simultaneously.

DPM 3 SE Composition Center.

Finally, multiple user samples can be mapped across an entire keyboard (i.e. an entire drum kit can be mapped with an entire orchestra of samples under one keyboard without patch changes.) Memory capacity is the only limitation.

SPECS AND DESCRIPTION

Manufacturer:	Peavey Electronics Corporation
Contact:	Doc Adkins 711 A Street Meridian, MS 39302-2898 601-483-5365 Fax: 601-484-4278
Model(s):	DPM 3 SE Synthesizer keyboard $2,999 DPM SX 6-bit sampling expander $349.99 Mega Sample RAM board Increases sample RAM to 1Mbyte $159.99 DPM SP 16-bit sample playback module $999.99

Mike Oxlong is an independent MIDI system consultant and free-lance writer.

The Peavey DPMSX sampler/expander.

SPEAKING OF MEMORY

DPM 3 SE factory units are equipped with 64K of user RAM. Installation of the optional Mega Sample RAM Board upgrades the original 64K of static "user-definable" samples to professional standards compatible (read: longer sampling time). With 100 ROM- or RAM-based sounds, a Mega Sample RAM Board provides the user 13.6 seconds of 16-bit samples at 38.4kHz. Internal playback speed of the 3 SE is 38.4kHz.

Two varieties of memory upgrades are offered. Users can install eight, 28-pin, 32K chip sets or eight, 32-pin 128K chip sets, yielding a total of either 256kbytes or 1Mbyte of static RAM. Internal programming structure (synthesis architecture) and RAM allotment is further enhanced by upgradeable factory revisions of software versions. The current software revision is Rev. 2.3.

The 3 se offers other storage media via an on-board 3.5-inch double-sided, 720kbyte disk (which can be formatted by the DPM or by any MS-DOS or Atari computer). More storage happens when a 32K ROM card is inserted into the rear panel memory cartridge slot which doubles 3 SE onboard programs to 200. (Storage is also available via porting MIDI data to an external hard disk drive.) The 3 SE's SysEx dump feature permits upload and download of file libraries in seconds.

VOICE EDITING

Many users desire to access and edit raw sample data or the DPM SE's resident ROM "waves" according to needs. The 3 SE does this through its voice editing section.

The 3 SE can store up to 48 raw samples and 32 multisamples in sample RAM. A total of 105 internal ROM waves are the building blocks of the synthesis process.

Voice editing is achieved through a synthesizer-like control matrix. Waves and samples can be modified through two oscillators (one wave each), two digitally controlled amplifiers (one per oscillator), a low-pass filter, four envelope generators, two low-frequency oscillators, an output amplifier stage and an output effects router (similar to a console's effects send). Waves and samples can also be combined using a "multi" option to layer sounds via keyboard ranges (splits), velocity and pressure.

SEQUENCING

Finished sound files next enter the DPM's sequencer. This digital MIDI recorder functions similarly to an external, computer-based sequencer. Depending on the application, it makes it possible to stay inside the machine until the final 2-mix. The sequencer can use five discrete drum kits, a 9-track recorder with automated mixing and level controls, and a digital onboard signal processing module whereby four effects can simultaneously process sounds.

Sequencing track commands emulate multitrack recorders. Such operations as record-ready mode, track selection, rewind, fast-forward, pause, stop and play transport controls are straightforward.

DATA CONTROL

Sound generation by the DPM 3 SE is managed by a simple soft-button user interface, front-panel accessed. Sound file menus are manipulated by the 12-button "voice edit" section located front and center above the keyboard. Menus can be viewed and edited by using a combination of "+" and "−" touch-buttons to select page menus increments or to fine-tune parameter values. Coarse parameter value adjustments are obtained by using either a linear slider or data wheel.

Peavey's 3-edition Composition Series library provides more than 100 rhythm patterns.

Interface layout is comfortably designed for sampling, editing and recording functions. The five sections are color-coded for easy identification. A 40×2 back-lit LCD serves as the main character display and features angle-viewing controls.

PERFORMANCE CONTROL

The 5-octave, velocity and pressure sensitive synth-action keyboard also edits sound files. Two mod wheels (one dedicated for pitch bending and the second assignable to any modulation source) enhance performance control. Footswitches and other MIDI controllers can be assigned via a back panel 1/4-inch jack and MIDI receptacles. A headphone jack has an independent level control. Two 1/4-inch jacks provide mono and/or stereo left/right unbalanced outputs.

MIDI IMPLEMENTATION

The DPM's MIDI capabilities define control of the unit's three MIDI ports by assigning MIDI transmit and receive channels. MIDI omni-, poly- and multi-control modes, along with complete or partial parameter filtering, are present. Four discrete multimode presets enable an external sequencer to record and drive as many as 16 separate sound files at once.

EDITING FX, DRUM MACHINES AND SAMPLING

DPM's master mode is used when editing and mapping the onboard drum kits, FX processors, user-tuning, external MIDI controllers (such as footpedal and breath controllers) and the sample RAM section. Drums, effects, tuning and controller selection are also edited to onboard sound files. The 3 SE differs from most music workstations in the sample RAM section. Here, users download samples from the SX sampling unit for more elaborate editing.

With the DPM 3 SE connected to a DPM SX (or connected to another SDS sampler) by MIDI cables at both in and out ports, the 3 SE can receive, load and save user samples to diskette and transmit compatible 16-bit samples to other DPMs or SDS units. SX specifications include full 16-bit delta-sigma A/D conversion, selectable sampling rates (16kHz, 24kHz, 32kHz, 38.4kHz, 44.1kHz or 48kHz, and line/level or microphone [with phantom] inputs).

The 3 SE is optimized for the SX. Beyond traditional trim and looping control, editing options include sampling frequency adjustments, threshold level, digital word length, sample record arm/start and sample dump which are all available at the front panel of the 3 SE.

The sample transfer rate is slow. Of course, the normal sample data transmission speed over any MIDI bus is hopelessly sluggish. Anyone working in time-is-money situations should save all samples to disc for immediate access.

OPTIONS, OPTIONS EVERYWHERE ...

In addition to the 3 SE's Mega RAM Sampling Board, the SX sampling front end expands up to 16Mbytes by using standard Macintosh 80ns or 100ns (SIMM) chips. Because the DPM 3 SE, as many other workstations, can only use of a maximum 1Mbyte, the SX offers an inexpensive, professional alternative to costly single-channel A/D converters.

For engineers or producers new to sound design, and/or pressed for time, an 8-edition library of factory supplied waves and samples enables user access to 100 unique sound files per edition. The library is read by the DPM's onboard 3.5-inch disk drive or cache card port.

Peavey's 3-edition Composition Series library provides more than 100 rhythm patterns, which will be helpful to novice sequencer users and those needing generic rhythm tracks. Experienced users, seeking to construct a strictly sample-based library, will fancy Prosonus' 8-edition Peavey set (SPARS Code: DDD). These fully digital samples were culled from Prosonus' stellar compact disc library and range from traditional orchestral samples to sound effects.

Although basic samples from Prosonus hover around or below 64kbytes, heavy users will opt for the full 1Mbyte option. Immediate access to Prosonus' sound files seems mandatory to anyone who's in it for the money.

For sample editing, the recommendation from Peavey designers is Turtle Beach System's

The Peavey DPMV3 Synthesizer Voice Module.

SampleVision 2.0. SampleVision overlaps the DPM 3 SE's onboard user sample editing screens. Based on intuitive graphical user interface (GUI), SampleVision offers superior visual editing, digital signal processing, and has drivers for various sampling keyboard devices. Both the DPM 3 SE and DPM SX models are supported by Turtle Beach's adherence to the MIDI SDS format.

For those not needing a 3 SE, Peavey's SX offers a logical alternative to expensive A/D converters. SampleVision and the SX together create a cost-effective digital sampling editor. If you've already dedicated a sampler for playback purposes, the SX becomes your recorder while SampleVision links the front end and existing equipment.

SPOT PRODUCTION AND MUSIC COMPOSITION

To really find out about the performance of the Peavey, we took the unit out for a real-life spin. In a typical session in a notable Chicago studio, The DPM worked well, as advertised. Background vocals and spoken trailer tracks were recorded to 2-track analog before sampling and assembly sessions. Once committed to tape, vocal tracks were then re-recorded by the SX sampler. Each vocal sound bite was transferred individually to the DPM 3 SE for digital editing (trimming, looping, balance and deletion of extraneous noises). This sent the talent home early and brought talent fees in under budget.

Once inside the 3 SE, samples were assigned to particular sound files. Because each sound file can be modified as if it were a non-sampled waveform (i.e. a musical instrument) the engineer could then modify the amplitude time and levels of certain background vocals to better suit the producer's requirements. Filtering of the spoken trailer by the engineer eliminated brittleness of the narration. Panning each vocal sound file into a discrete position broadened the stereo sound field.

For creation and sequencing the musical portion to the vocal samples, the producer chose to track the electronic keyboard and drum directly from the DPM 3 SE's internal sound files to the internal sequencer. Keyboard talent was ready to assist in the creation of a simple drum groove (using two of the five 3 SE onboard drum/percussion machines) and the synchronization of slap bass and some lush keyboard pads.

A decision to fly previously loaded vocal samples over the musical sequence was made because of time constraints placed on keyboard talent. Rather than record separate vocal track(s) in the internal sequencer along side of the rhythm section sequence, the keyboardist "performed" the vocal sound files live directly to the 2-track mix. Because the DPM 3 SE allows sound file selection for performance, along with a sequenced pattern or song, this task was simple.

With multiple parts (combining musical sequence and vocals) needing to be mixed directly to DAT, a combination sound file represented all of the sampled vocal parts under one master soundfile (combination mode). Thus, all four vocal sound files appeared under the entire keyboard and eliminated any need to stop tape and switch sound-file memory locations.

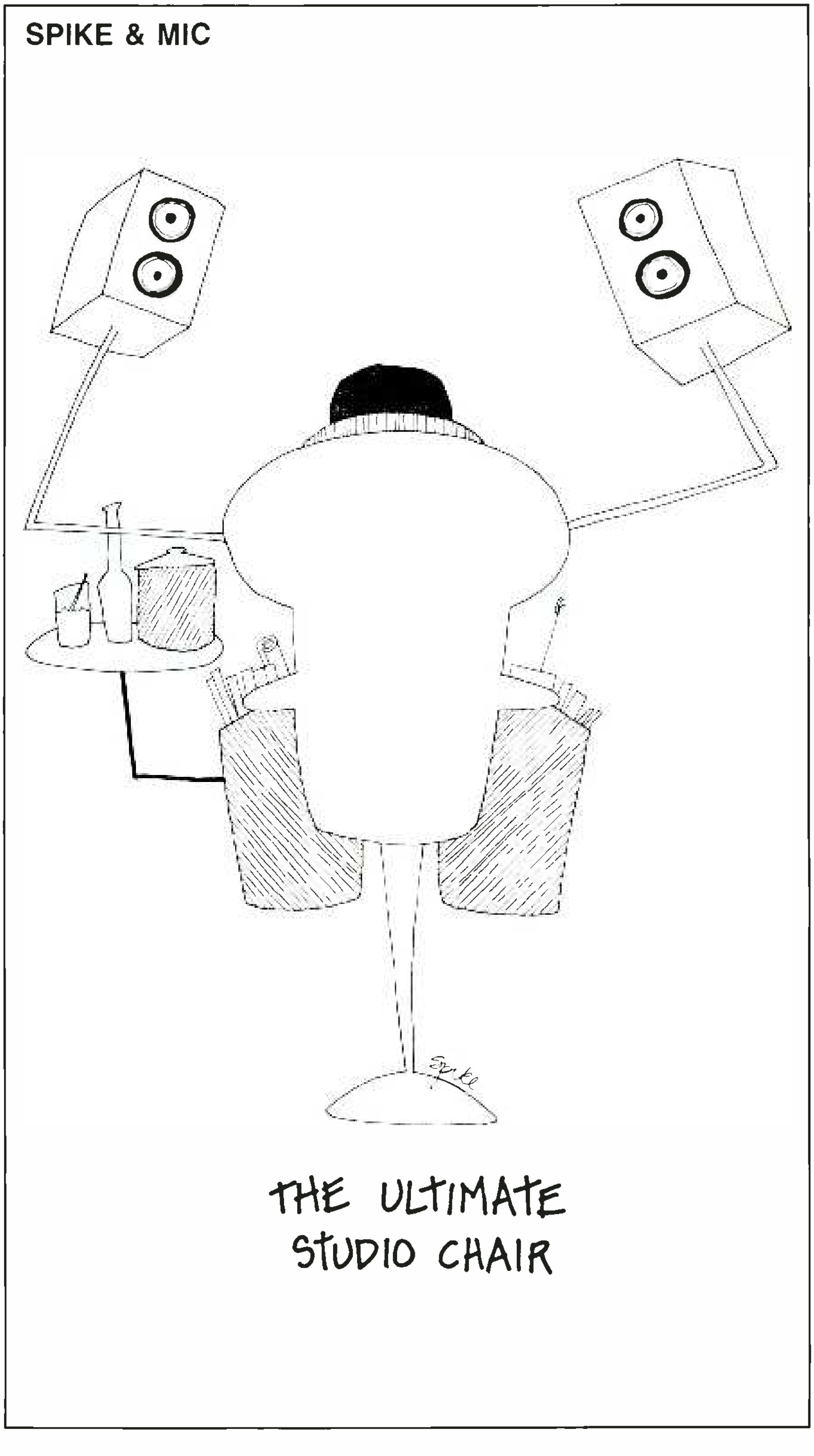
TO BUY OR NOT TO BUY?

Eventually, music/production worskstations will be as indispensable as the notebook-sized PCs now flooding the market. At press time, however, can't-live-without-it users are individuals who must remain mobile. Fixed studios most need workstation technology for second rooms and project development.

Portable keyboard sales are off nearly 20% for 1991, and several established competitors offer workstations. It's a buyer's market. With this in mind, it is fair to say that the DPM compares favorably with second- generation competitive units. Depending on the application and budget, the DPM 3 SE may match your needs, and studio.

Peavey's recent track record of intensive R&D and commitment to customers bodes well for potential users. The company is equipped to go the long haul. ■

Circle (100) on Rapid Facts Card



First Look

Things with Time Code

By Laurel Cash-Jones and Fred Jones

This is our favorite kind of product: one that solves a problem that arises from the omission of a *necessary* feature by one manufacturer and is built by another manufacturer to help out all of the people who needed it in the first place. It is toward this happy solution that Cipher Digital introduces the CDI-825. Before we delve into this product, let us consider the other product that it works in conjunction with. (Bear with us, we'll straighten it out in a moment).

In case you are unaware, Sony has been trying to introduce Hi-Band 8mm to the professional, industrial and consumer market for a few years. This format has several advantages over VHS, not the least of which is the fact that they have included a time code track plus stereo digital audio tracks on the professional versions.

However, don't get to excited about the time code track. Unfortunately for us, there is no way to get to the time code other than using the serial interface that is meant to work with a video editing system.

By the by, the digital audio tracks are currently 8-bit with a 32kHz sampling rate, but a new version (at least for consumers) is expected early next year with 16-bit and either 44.1kHz or 48kHz sampling rates.

If you're reading this at Sony Pro, a Hi-Band 8mm deck with time code in and out, and high quality audio would do *very* well as a professional alternative for home/MIDI studios to use instead of a 3/4-inch or VHS deck. If you're reading this in the consumer division, we would also like to have one without the time code track for our system at home. Just a hint.

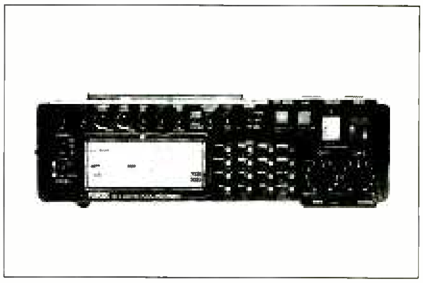
Back to reality. The Sony model EVO-9800 has time code (sorry, out only) available on it's 9-pin RS-422 control port. Those clever folks at Cipher Digital decided to make a serial-to-longitudinal time code device so that those of us who do not own a video editing system can take advantage of this feature.

The CDI-825 operates in either of two modes: stand-alone or monitor mode. In the stand-alone mode, the 825 requests the time code information from the deck each frame and outputs it in LTC form. In the monitor mode, the 825 is inserted between the deck and a controlling device, such as a video editing system. In this mode the 825 does not request any data, it spies on the communication between the controlling device and the deck (not unlike the CIA). When the editor requests the time code data (typically each frame) the 825 also receives it and outputs the data as LTC, without affecting any other function of the editor, thus becoming transparent to the editor. The CDI-825 also outputs LTC at any speed, and compensates for "on-time" displays in time code readers by subtracting a frame from the code, thus enabling the reader to display the correct time.

Circle (101) on Rapid Facts Card

MORE TIME CODE THINGS

From Fostex comes the portable DAT deck we have all been waiting for. If you are into field recording for television or film and need to use time code, this is the 4-head DAT machine you *must* have. It's called the Fostex PD-2, and from its built-in monitor speaker and slate mic, to its extremely ergonomic design, we were unable to find any field-usable feature that was not included. Indeed, it has so many features, we are sure we are going to leave something out, but rest assured that Fostex did not.



Since we don't know where to start, let's take a look at the display. The large LCD shows all the usual things such as DAT absolute time and SMPTE/EBU time code, but it also shows you the pre-selected operational settings, such as frame and sampling rate, and sync source.

As far as getting in and out of the PD-2, in addition to all of the normal analog and digital audio connections, it has BNC-style input and output connectors for both video and external sync in with video composite, 25, 29.97, 30fps frame, 24, 25, 29.97, 30fps field, 48, 50, 59.94, 60 field/second rates standard. Digital word sync in accepts three sampling frequencies (44.1, 48, and 44.056kHz). It will also record in each of those frequencies. Of course, an internal time code generator that can generate code or jam sync to any external source or regenerate actual camera time code is included.

On the input stages mic/line switching with a 30dB pad and phantom powering of 48V and T12-type voltage is standard, and a 3-position steep bass cut filter is integral, with settings at 40Hz, 80Hz, and 400Hz, plus Channel 2 has a phase reverse switch for stereo coherence.

The indexing and slating features are also unique. Take numbers can be entered manually or automatically; You can slate via the built-in mic or use just the tone generator. You can also mark the PCM errors (they are recorded as PNOs from 700), input overload or time code dropout errors. Errors can be marked so that you can search by them, or you can search by time, or index points) The PD-2 can also be used in an edit suite using the RS-422 9-pin port to talk directly to most existing editing systems. We could go on, but space does not permit.■

Circle (102) on Rapid Facts Card

Laurel Cash-Jones is R•E•P's editorial consultant and a Los Angeles-based free-lance writer. Fred Jones is an audio industry observer and a Los Angeles-based free-lance writer.

RANE LINE TRANSFORMER

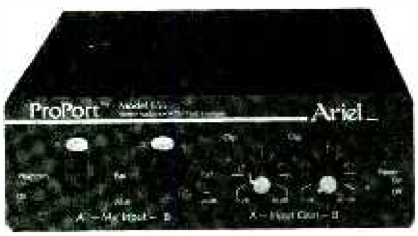
Rane is now offering the FLT 22 Line Transformer module, which provides two low-distortion, wide-bandwidth nickel core audio transformers to convert unbalanced line-level signals to isolated balanced outputs. The FLT 22 is housed in a Flex Series HR format chassis and can be expanded to four channels via the Option 44 expander board. FLT 22 input/output connectors are terminal strip and the unit may be rack-mounted horizontally, vertically or left as a stand-alone shelf-top unit. Where space and/or budget is tight, the Option 44 card may be used as-is, without the HR chassis — the terminal strip is mounted right on the PC card.

Circle (106) on Rapid Facts Card

ARIEL DIGITAL AUDIO INTERFACE

Ariel Corporation has debuted the ProPort Model 656, a self-contained digital audio interface that brings recording studio quality analog audio to ISA/EISA, Sun, VMEbus, Macintosh, Hewlett-Packard, and NeXT computers via the DSP Port serial interface.

ProPort provides two channels of 20-bit, 8-times-oversampled digital-to-analog conversion with a selectable sample rate from 5kHz to 96kHz, for any signal I/O including speech processing, laboratory data acquisition, signal generation and speech research.



ProPort employs 16-bit oversampling technology for the input converters, electronically balanced microphone and line-level inputs, switchable phantom microphone power, peak reading level indicators, continuously adjustable gain controls with 60dB range and over-voltage/surge protection. Analog outputs are driven by active-balanced, low-impedance line level amplifiers.

Circle (107) on Rapid Facts Card

LEXICON OPUS UPGRADE

Lexicon is now shipping OPUS Version 3.0 featuring extensive enhancements to the OPUS digital production system. System enhancements affect both hardware and software, and make OPUS the industry's only fully automated digital mixing console, as well as the only system to integrate every major audio post-production function.

Circle (122) on Rapid Facts Card

JBL STUDIO MONITORS

JBL's 4200 series 2-way studio monitors strike a radical new outward appearance in the quest for superior console-top monitoring. The 4206 features a 6.5-inch woofer, and the 4208 an 8-inch woofer and were designed specifically for use in the near field. The Multi-Radial sculptured baffle directs axial output of the individual components for optimum summing at the most common console-top listening distance, approx. 1 to 1½ meter (3 to 5 feet). The Multi-Radial baffle also positions the transducers to achieve alignment of their acoustic centers resulting in superb imaging and greatly reduced phase distortion.

The curved surface of the injection-molded ABS baffle serves to direct possible reflection of the shorter wavelengths away from the listening position and reduce baffle diffraction distortion. The ducted port was moved to the rear of the enclosure to further reduce distortion. Vertical alignment of the transducers across the baffle center produces natural mirror-imaging.

Circle (105) on Rapid Facts Card

STEWART ELECTRONICS POWER AMP

Stewart Electronics has added the model PA-800 power amplifier to their amplifier arsenal. A single rack-space unit, the PA-800 employs the same technology as the larger PA-1200. The PA-800's high efficiency "Switch Mode Power Supply" delivers significant increases in the amp's efficiency, and simultaneously allows most of the amplifier's major components to be downsized, saving space, weight and cost. Typically, six PA-800's can co-exist on a standard 20-amp circuit. Each PA-800 delivers 400W per channel into a 2Ω load.

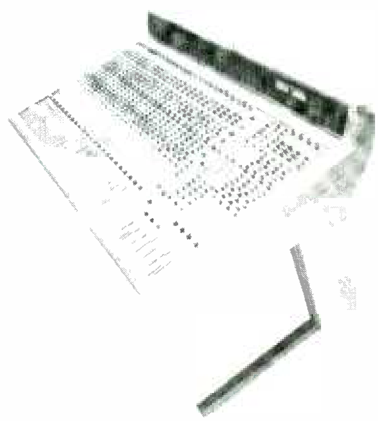
Circle (115) on Rapid Facts Card

SOUNDCRAFT IN-LINE CONSOLE

The Sapphyre is Soundcraft Electronics' newest recording console. Designed for the mid-level recording market, Sapphyre's performance specifications and design make it ideal for recording and post production applications. The console is available in 20-to 44-input versions with or without integral patchbay.

Each I/O module incorporates individual noise gates with an advanced 4-band EQ design, splittable between the two signal paths. The dual line input option enables increased input capability for effects returns or virtual tracks. A combined I/O module gives access to dual signal paths, one for monitoring and the other for recording, with sub-grouping and routing architecture enabling different modes to be easily configured and controlled.

Sapphyre boasts eight sub-group buses that can be used as virtual patchcords, allowing signals to be re-routed to any tape channel directly, or to an input, and then sub-grouped to tape. Frame sizes are 20, 28, 36, or 44 I/O modules, each with six stereo effect returns.



Circle (109) on Rapid Facts Card

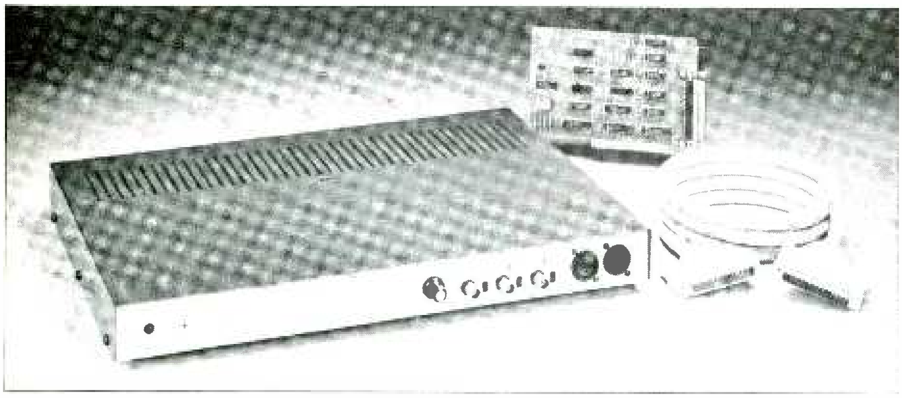
Cutting Edge

TECHRON INTERFACE

Equipped with the same powers as the TEF 20, Techron's TEF 20HI includes a high-speed interface that allows the device to be used as a digital audio recorder. This same HI interface also permits third-party software programs to utilize the TEF 20 DSP chip to control the entire system.

The TEF 20HI's high-speed interface provides the user with a wide variety of tools and functions not usually associated with TEF, such as the ability to become a top-notch digital oscilloscope, real-time analyzer, filter generating unit, and data capture and display/analysis device.

Circle (110) on Rapid Facts Card



MARK OF THE UNICORN TIME PIECE

Mark of the Unicorn's new IBM PC/compatible version of the MIDI Time Piece is a multi-cable MIDI/SMPTE interface for computer-based music production systems. MIDI Time Piece features eight independent MIDI Input/Output cables. Each cable has 16 MIDI channels for 128 MIDI channels per unit. MIDI Time Piece also has complete MIDI merging, routing, channelizing and event muting capabilities. The device functions as a standalone merger/mapper when the computer is turned off. An advanced MIDI/SMPTE reader/generator/converter for tape synchronization completes this three-in-one device.

This IBM version features an 8-bit interface card that allows connection of up to two MIDI Time Pieces to the computer for a MIDI network with 256 channel and 16 MIDI inputs and outputs. Two separate computers can share the same MIDI Time Piece network.

Circle (116) on Rapid Facts Card

API CONSOLE

Audio Products introduced the fourth in a new series of discrete consoles during the 1991 New York AES Show. Configured with up to 128 inputs and 49 buses, this console's new customized features include an automated send module capable of assigning individual sends (available in configurations of 8, 10, or 12 per module) to pre or post on either the channel's large or small fader.

Sends can be individually muted through the GML Automation Environment. The console's "Touch Reset" control creates resettable switch settings on the entire console by the main computer.

Circle (108) on Rapid Facts Card

CREATIONS TECHNOLOGIES MIDI DRIVER

Anatek's MIDIMatch line driver is designed for people needing to control MIDI lighting systems, sequencers or computers up to 4,000 feet away. MIDIMatach data can be sent through any two conductor-shielded audio cable, and the signals can be treated like an audio signal and sent through a patchbay with no signal loss or MIDI delay.

Each unit has two completely independent transmit and receive circuits that can be used to send and receive two different data streams. MIDIMatch signals are immune to noise and will not interfere with any other signal carried on adjacent lines. The system consists of two identical bi-directional units; one for each end of the cable, and two 9V power supplies.

Circle (125) on Rapid Facts Card

SONIC SOLUTIONS-PREMASTER CD

Sonic Solutions has unleashed advanced technology for complete tapeless mastering of audio CD and CD-ROM. Sonic Solutions' PreMaster CD offers economy, reliability and the ability to quickly check the master before it goes to the plant.

With the Sonic System and a CD Maker or CD Printer, a record company or mastering studio can record a finished program onto a PreMaster CD (PMCD). In addition to the audio program and a table of contents for standard CD player, the PMCD contains information required by the code cutter such as precise timing information for all track starts and indexes, plus data relating to copy prohibit, emphasis, ISRC code, etc. At the CD plant, the MasterMaker, a Sonic-System/CD Maker combination outfitted with special software, reads back and relays data to the code cutter which in turn writes the glass master.

Circle (113) on Rapid Facts Card

TIMELINE CONTROL

The TimeLine Console Control Unit (CCU)is a miniature keypad that mounts directly into standard Neve, SSL, Euphonix, and other consoles. The CCU operates the TimeLine System Supervisor multiple machine controller which interfaces to standard console automation software with no changes or updates required. Utilizing Lynx Time Code Modules, the CCU controls up to six analog or digital audio tape recorders, VTRs or sprocketed film transports. All data communication is processed by the TimeLine system; all machines are operated directly by the console automation.

Through the CCU any transport may be designed as the master without switching cables. The CCU system offers variable speed control of the master for pitch changes of an entire synchronized machine group. An optional jog/shuttle wheel is available.

Circle (112) on Rapid Facts Card

ELECTRO-VOICE ACTIVE CROSSOVER

Electro-Voice has introduced the EX-24, a stereo 2-way, mono 3-way crossover designed to maximize biamped and triamped system performance.

The 1-rack unit EX-24 offers 12 selectable crossover frequencies per channel. Frequency settings are 80Hz, 125Hz, 160Hz, 250Hz, 500Hz, 630Hz, 800Hz, 1250Hz, 1600Hz, 2500Hz, 500Hz, and 6300Hz.

I/Os are $^1/_4$-inch balanced or unbalanced and XLR balanced. Discrete channel controls over low and high output levels, polarity switches and on/off output switches serve to ease system setup.

A switchable horn equalization circuit is present to flatten system response when using constant-directivity horns. Infrasonic filtering (–3dB at 30Hz) and stereo or a mono subwoofer option enhance the low pass circuitry.

The internal power supply may be configured for global applications. Included is an IEC connecter with detachable ac line cord.

Circle (114) on Rapid Facts Card

AUDIO TECHNICA STEREO MICROPHONE

Audio Technica has introduced the AT822 One Point X/Y stereo condenser microphone designed specifically for DAT and high performance cassette recording. Mono compatible, the AT 822 is also intended for television, FM and field recordings. Inside of the AT 822 are a pair of wide-ranged and close-matched cardioid condenser elements delivering natural response across an arc of 170°

The high output stereo AT833 terminates its standard cord with two mini plugs threaded inside a pair of 1/4-inch phone plug adapters. The AT822 operates on a standard 1.5V AA battery and includes a switchable low-cut filter, windscreen and camera shoe mount adapter.

Circle (135) on Rapid Facts Card

APHEX 9901 PARAMETRIC EQ MODULE

The single channel 9901 Parametric EQ from Aphex is intended for live sound applications as well as recording, film sound and broadcast environments. As the latest addition to Aphex's 9000 Series modular processing rack system, the unit offers three overlapping bands of EQ with a peak/shelf filter on each band.

Each adjustable band has ±15dB of boost or cut with transformerless servo-balanced input and output circuitry. Like the other Series 9000 modules, the 9901 fits into both the 9000 rack and the dbx series 900 modular frame.

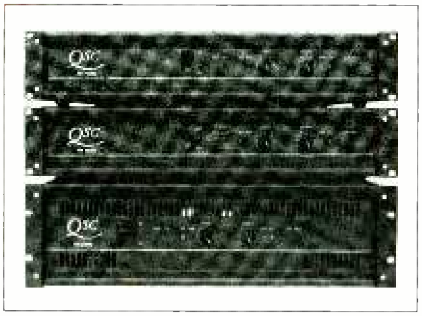
Circle (126) on Rapid Facts Card

CROWN AMPLIFIER

From the output end, the first in the latest generation of Micro-Tech amplifiers, Crown's Macro-Tech 3600 VZ produces a maximum of 3,600W of power in a 2-rack-space frame. Operable in stereo, bridged mono, and parallel mono modes, the 3600 VZ claims a 105dB (A-weighted) S/N ratio at full output, with 26dB of gain. Frequency response rates at ±0.1dB from 20Hz to 20kHz at 1W, while THD was measured at ±0.05 from 20Hz to 1kHz with lineal increase to 0.1% at 20kHz at full output.(Evaluations were made in stereo mode with both channels driven into an 8Ω load.)

Circle (127) on Rapid Facts Card

QSC POWER AMPLIFIERS



QSC Audio has released their new MXa power amplifier series. The "a" designation behind the MX1500a and MX2000a models signifies increased power and lighter weight versions of the previously released MX1500 and MX2000. Additional improvements include automatic fan speed control, an input slot for additional connecters, indicators and both active and passive input accessories. Prices have not increased. The model MX1000a is an all new product. Stereo, 8Ω output ratings for these units are 450W per channel for the M2000a, 250W per channel for the 1500a and 250W per channel for the 1000a.

Circle (137) on Rapid Facts Card

HYBRIDS ARTS RECORDER

Hybrid Arts, Inc. has begun shipping the Digital Master, an affordable direct-to-disk recorder/editor. The complete systems includes a CPU, monitor and Mouse, 105Mbyte hard disk, A/D and D/A converters, MIDI, SMPTE interface, and software. Software features include comprehensive graphic editing functions and various playback functions such as non-destructive editing, a sound effects cue page, over an hour of continuous recording time, and up to 14 hours of sound on line. The hardware includes an S/PDIF (AES/EBU compatible) digital audio interface and a SCSI port for connection of common hard disk drives.

The Digital Master is a stereo, direct-to-hard disk audio recording system for the Atari 1040STE (with 4Mbytes of RAM), Mega4ST, and Mega4STE with 16-bit, 64× oversampling A/D converters and dual 18-bit, 8× oversampling D/A converter. It also offers selectable 48kHz, 44.1kHz, 32kHz, 31kHz, 25kHz, 22.05kHz, and 15.25kHz sampling frequencies. Digital I/O is via S/PDIF and AES/EBU (RCA Connecters). Support is provided for 24, 25, 29.97 and 30fps SMPTE time code. Options include time compression, real-time and offline digital filtering. The Digital Master has a frequency response of 10Hz to 20kHz (0.1dB), greater than 96dB dynamic range, a S/N ration of greater than 90dB (full-scale, at 1kHz, A-weighted), and less than 0.02% THD plus noise. Digital Master works with an SCSI hard disk with a minimum transfer rate of 350kbyteB/second and a maximum seek time of 50ms.

Circle (117) on Rapid Facts Card

KLIPSCH LOW FREQUENCY LOUDSPEAKERS

Klipsch and Associates has also begun shipping the K-1200, K-1500, and K-1800 Series, professional-level 12-inch, 15-inch and 18-inch woofers, marking the company's entry into the "raw frame" component marketplace. High frequency drivers will soon join the K series.

K series speakers can handle 300W of continuous pink noise from 40Hz to 2kHz for eight hours, with peaks to 3kW. The 12-inch speakers have 77 ounce magnets, while the 15-inch and 18-inch speakers have 96 ounce magnets and employ 3-inch Kapton voice coil forms. K-1200 speakers are 8Ω designs, and the K-1500 and K-1800 speakers are produced in both 4Ω and 8Ω. The 15-inch and 18-inch speakers are also available in models for small bass reflex enclosures as well as horn enclosures.

Circle (131) on Rapid Facts Card

LEMO CATALOG

LEMO USA has released its newest catalog for connector specifiers of audio and video equipment. Designed to include the most up-to-date technology of the audio/video industry, the catalog features coaxial and triaxial connectors for audio, video, and TV camera applications; multicoaxial connectors for the state-of-the-art HDTV industry; and 50Ω or 75Ω connectors combining coaxial, triaxial and signal (low voltage) contacts for these applications. Connectors come in a variety of sizes and shell styles including a sealed version.

Also detailed in this catalog are LEMO's newest patch panels for audio and video applications. All LEMO connectors feature a self-latching, push/pull-type design to ensure reliable connections and avoid signal interruption by accidental pulling of the cable.

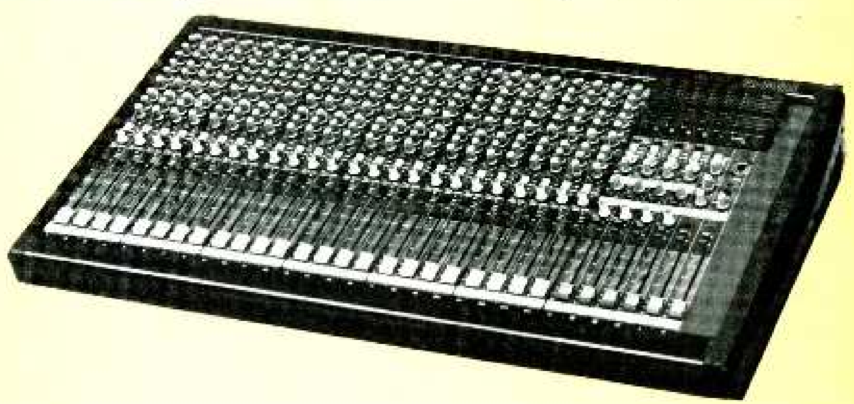
Circle (124) on Rapid Facts Card

RAMSA MIXING CONSOLES

The Ramsa WR-S4400 console series offers 12-, 16- and 24-channel professional 4-bus mixer consoles featuring professional length faders (100mm), two selectable inputs per channel, individually switchable 48V phantom power, flexible 3-band EQ with sweepable midrange and layout similar to Ramsa's popular WR-S840 series of concert desks.

WR-S4400 outputs feature four main groups, plus left and right stereo masters from channels or groups and four aux sends. To increase the available Aux groups, Ramsa added a D-out switch and output to each input channel. The switch routes the channel's signal through its Aux bus control, and off Aux 1 bus to its direct output, creating up to 15 aux sends on the 12-channel board, 19 on the 16-channel and 27 on the 24-channel without affecting any other channel operation. The result is an aux group multiplier that operates only per input channel and allows the use of individual effects like voice limiters or reverb devices without taking up one of the conventional four aux groups.

Circle (129) on Rapid Facts Card



INTERFACE MIXING CONSOLES

Interface, a new series of modular mixing consoles, is now available from several Mark IV audio companies. The Interface desks are being manufactured under Mark IV's "multi-brand concept" and will be marketed by Altec Lansing, DDA, Dynacord and Electro-Voice.

Available in 8-, 16-, 24-, 32- and 40-channel mainframes, Interface features include five LED level indicators on each channel, padded mic inputs, Aux sends which are switchable to direct channel outputs and a pre/post switch on Aux sends one and two. The group module is equipped with extensive switching for PA or recording applications.

Four group mixing buses allow the use of up to four group output modules. Six Aux buses are also provided, giving six additional mixes with master level controls. Optional input and output transformers are available to isolate the electronically balanced XLR connecters. The 8-channel model is also available as a rack-mount.

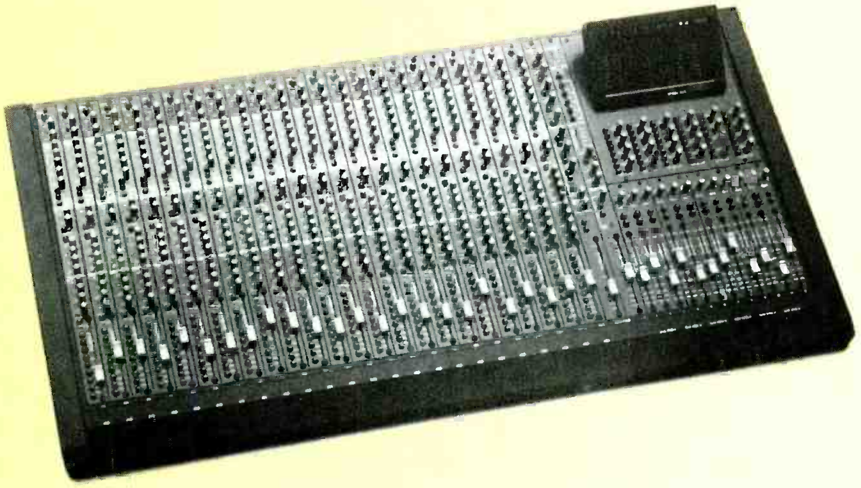
Circle (142) on Rapid Facts Card

FOSTEX MIXING CONSOLE

The new Fostex 2412 mixing console is a 24×12×2 recording console that can manifest two simultaneous stereo mixes via two complete mixing paths. Both mixes have EQ, Solo, Aux sends, and Foldback/Cue sends. This configuration results in the equivalent of 60 available inputs during a stereo mix.

Automated MIDI muting, stereo in-place Solo, six Aux returns and split equalization (Hi, Lo and two bands of sweep) complete the a 22-inch deep package.

Circle (138) on Rapid Facts Card



MARION SYSTEMS SCSI

Marion Systems recently confirmed availability of the MPC-SCSI hard disk SCSI interface for the Akai MPC60 and MPC60-II. With the MPC-SCSI and a Macintosh-compatible hard disk, up to 780Mbytes of on-line storage can be accessed. Load and save sequences are four times faster than the floppy equivalent. The MPC-SCSI has been tested with multiple Macintosh-compatible drives, including Syquest, Conner, Quantum and Seagate.

The MPC-SCSI consists of a small circuit card and associated cables. Installation requires opening the MPC60, installing the new software ROMs, the circuit card, and attaching two cables.

Software for the MPC-SCSI was designed and written by Roger Linn and his team of engineers. Other than the hard disk portion, the software is identical to Akai's latest software revision for the MPC60.

Circle (121) on Rapid Facts Card

MCCAULEY SPEAKERS

McCauley Sound of Puyallup, WA, has announced a new line of extended low range loudspeakers available in 12-, 15- and 18-inch sizes. Designed to handle extremely high-power applications (400-450W RMS) with minimal distortion and breakup, all three speakers share the same field-serviceable magnet assembly. Removal of three allen-head screws enables users to separate the magnets from the baskets to inspect and service the speaker, thus reducing down time and even the need for reconing. The interchangeable magnet will fit any size basket, allowing speakers to be changed to accommodate user needs.

The 18-inch 6254 provides 450W RMS with a frequency response of 20Hz to 800Hz. The 15-inch 6242 is rated at 450W RMS, with a frequency response extending from 1.2Hz down to a clean 25Hz. The 12-inch 6232 model takes low frequencies from 40Hz to 2kHz.

Circle (128) on Rapid Facts Card

SOMICH ENGINEERING HEADPHONE AMPLIFIER

The new HPX high performance headphone amplifier from Somich engineering was designed to drive dynamic headphones without the coloration inherent to most high current headphone amplifiers. The HPX circuit topology uses a minimalist approach, relying on highest quality monolith components to recreate the loudspeaker sound stage, to provide 20dB of class-A voltage amplification.

Circle (133) on Rapid Facts Card

Cutting Edge

AUDIO PRECISION FASTEST SYSTEM

Audio Precision has introduced the FASTest system which can acquire conventional analog measurement results in seconds. The FASTest program for the System One+DSP and System One Dual Domain can characterize an audio device, system or channel for frequency response, distortion and noise, in a few seconds without compromising accuracy or resolution. Using FFT analysis of a composite multi-tone test signal, the relative amplitudes and frequencies of each discrete signal are analyzed and compared to yield these multiple measurements.

Circle (134) on Rapid Facts Card

ALPHA AUDIO PORTABLE BOOTHS

Alpha Audio Acoustics is shipping its new Audio Seal Portable Sound Booths. The booths are made of flexible panels constructed from the Audio Seal barrier and quilted fiberglass absorber combination blankets. Used as an on-location sound and video booth, the Audio Seal product is durable, yet easy to construct and dismantle. A secondary application for the booth is as a residential practice studio. The units have a steel frame and are assembled using component parts and velcro fasteners. These booths have a Standard Transmission Coefficient (STC 29) Rating and are Class I fire rated for flame spread and smoke density.

Circle (120) on Rapid Facts Card

HYBRIDS ARTS RECORDER ADAP IV

Hybrid Arts has introduced the third generation ADAP IV into the multitrack, direct-to-disk, digital recording/editing market. ADAP IV's standard hardware consists of four channels of high quality analog input and output, 16-bit, 64× oversampled delta-sigma A/D and 8× oversampled dual 18-bit DAC's, 4-channel digital I/O, a custom DSP module, a CPU with mouse, keyboard, and monitor, MIDI and SCSI ports and a built in SMPTE interface for true "chase-lock" operation. Up to seven hard disks can be connected for a total recording time of over 12 hours.

Circle (118) on Rapid Facts Card

API COMPRESSOR/LIMITER

API has also unveiled the new 525B, a higher evolution of the original 525 compressor/limiter. Continuing the longstanding API philosophy of building around the original circuit, the 525B will still only have two discrete op-amps in its signal path. The new compressor/limiter is enhanced with a gating function, a frequency sweepable de-esser and will be available for use in the 500-B4 Lunchbox, the 500-VPR, and 500-HPR power racks and the API Discrete Series Console.

Circle (123) on Rapid Facts Card

DRAWMER QUAD GATE

Drawmner is now shipping the DS404 Quad Noise Gate featuring "program adaptive" circuitry capable of handling an extremely wide range of program input signals. A hard/soft gating switch is a key part of the adaptability of the DS404. In "hard" position the DS404 provides ultra fast response, while in "soft" mode the unit assumes expander functions. Gentle release characteristics complement the "soft" gate mode. The DS 404 also exhibits frequency sensitive gating and a slave function when linked channels are required.

Circle (136) on Rapid Facts Card

CYCLONE SYSTEMS SIGNAL INJECTOR

Faults in audio equipment can easily be located in almost any location using the YIBBOX. New to the American market, the ingenious YIBBOX provides 400Hz outputs at 0dBm, –20dBm, –30dBm and –50dBm to check broadcast, microphone and keyboard feeds. It can also test loudspeakers and headsets. Portable, the YIBBOX weighs seven ounces and is powered from an internal 9V battery with a life of 24 hours continuous operation.

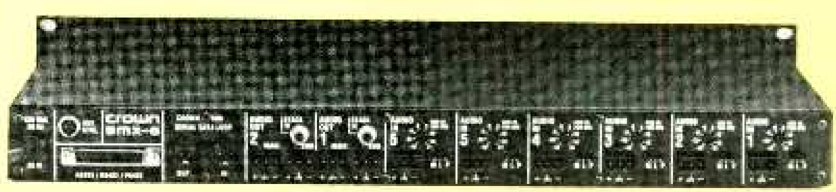
Circle (139) on Rapid Facts Card

CROWN MULTIPLEXER

Sensing technology is the major addition to Crown's new SMX-6 multiplexer. Based on the design of its predecessor, the MPX-6, this unit houses the power and intelligence to switch and route each of its six inputs, two summed outputs, and two relay-switched output buses. The SMX-6 is also capable of integrating with Crown's IQ System 2000 for digital control of time delay, equalizers, etc.

The inclusion of eight level detectors on both the units' input and output stages provides the next level of real time system status displays, such as monitoring of voltage and current across loudspeaker lines and summed representations, in bar graph form, of all input levels.

Circle (132) on Rapid Facts Card



CREATIONS TECHNOLOGIES HARDRIVES

The Anatek DS series of rack-mount hard drives are designed to overcome data storage space limitations by combining a large, fixed hard drive with a removable media drive in the same chassis.

This combination enables storage of CD-quality digital audio on the fixed disk which can be transferred to a removable media for transport or archival. The data remains in unfinished form as it is shuttled on-and off-line from the faster, fixed hard drive. Work may resume from precisely the same point once reloaded.

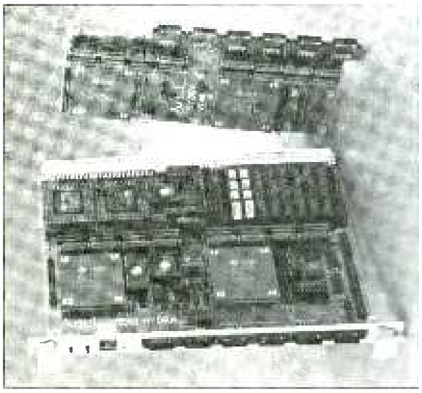
Anatek DS features for the professional recording environment include: worldwide power supply, dual fans that may be disabled for absolute quiet, high quality ac spike and surge protection for both the drives and auxiliary outlets, and proprietary Macintosh software for partitioning, password protection, spanning and automatic backups.

Preconfigured DS systems range from a 101Mbyte fixed hard drive and 44Mbyte removable hard drive for approximately 13 minutes of stereo to a 1Gbyte fixed drive and 1.3Gbyte DAT for a total of four hours of stereo sound storage. CD ROM drives are also available for access to pre-recorded sound libraries.

Circle (119) on Rapid Facts Card

AREIL DATPORT

Ariel's new DatPort is a self contained digital audio interface that can be used with any computer or DSP processor board that has serial DSP capabilities compatible with the NeXT DSP port. DatPort accommodates standard 48kHz, 44.1kHz and 32kHz sample rates and also is capable of generating other sample rates through its on-board crystal generator. When connected to the DSP port of the Motorola DSP56001 chip, non-audio data is interpreted and formatted by software running on the DSP. For other chips, data is controlled by front-panel switches.



Circle (141) on Rapid Facts Card

STELLAVOX D/A CONVERTER

Stellavox has collaborated with Swiss sister company Goldmund, to produce the new Stellamode D/A converter. The unit can convert from an AES/EBU or S/PDIF input. It also includes an AES/EBU output, so the Stellamode can be inserted into an existing AES/EBU loop without breaking or changing any connection. Analog outputs can be adjusted between 0,+4dB and +6dB and in polarity. To keep pace with technological evolution, the Stellamode's digital interface and converter-filter are molded in pluggable modules which can be upgraded as required.

Circle (143) on Rapid Facts Card

SPIKE and MIC

CUTTING EDGE

Continued from page 63

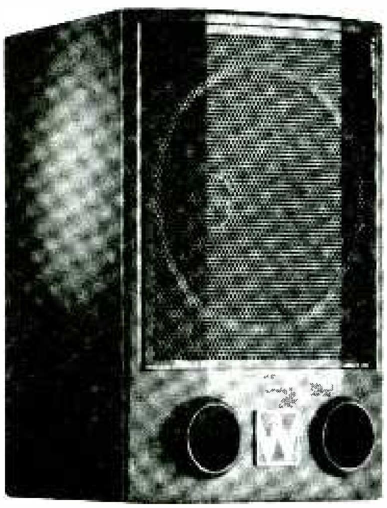
WOHLER TECHNOLOGIES LEVEL METERS

The new MSM series audio level meters from Wohler Technologies provides accurate monitoring of levels for up to 20 mono or 10 stereo audio sources in a single rackspace unit. The array of 10-segment LED bar graph meters, each with green/amber/red signal level indication, assures at a glance recognition of potential problems with missing signals, mismatched levels, or input overloads. "Zero" calibration level and VU or PPM ballistics are individually selectable on each meter.

Circle (130) on Rapid Facts Card

WHARFEDALE LOUDSPEAKERS

The new British-made Wharfedale Force 9 loudspeakers are now available in the U.S. from Optimum Audio of Stamford, CT. The Force 9 features a newly designed SMS co-axial driver with a 12-inch silicon impregnated paper cone and a 1-inch titanium compression driver sharing a common magnet in a trapezoidal cabinet. Extended bass response is achieved by two front firing tuned ports.



Force 9 drivers are protected from overload by two temperature coefficient devices with indicating LEDs flashing on front cabinet. Sensitivity is rated at 98dB at 1 meter on axis, 1W pink noise, a maximum SPL of 122dB at 1 meter continuous with 60° vertical and horizontal coverage. Power handling is 250W and frequency response covers 70Hz-20kHz at −6dB.

Circle (140) on Rapid Facts Card ■

DIGITAL DOMAIN

Continued from page 23

STANDARDS AND PRIDE

Some of you are probably thinking, why don't all of the companies standardize on the same terminology to eliminate all the confusion? While it would certainly be nice if workstation manufacturers shared a common command set, I don't believe it will ever happen. One reason is pride. Some of these companies are intense rivals, who would rather die than adopt the same terminology their competition uses. Another reason is that standards often create mediocrity.

Hopefully, when manufacturers see how they stack up against each other, they will rethink their choice of language to make sure it's concise and easy-to-understand. Unfortunately, it's hard to change horses in the middle of the stream. I've talked with programmers who feel it is important to be consistent with their terminology, even when it contradicts programming guidelines or is just plain confusing. I understand their fear of alienating existing users, but digital technology is still in its infancy. The time we have behind us is small compared to the time we have ahead of us.

THE SUE SYNDROME

Most graphic user interfaces (GUIs) use icons, pull-down menus and a mouse. GUIs have slowly evolved over the last 20 or so years. No one company or person can take the credit for the concept. It's unfortunate that the 'look and feel' lawsuits, waged by the software companies, have gained such widespread attention, because they discourage commonalty between software. Imagine if 30 years ago the courts had ruled that Ampex was the only tape machine manufacturer that could use the terms fast forward, rewind and play on their tape transports. That situation would be no more foolish than some of the debates going on in the software/hardware world today. Apple Computer's real competitor is not IBM or Microsoft, it's a pad of paper and a calculator. There are still millions of people who have never used a computer. I say, if someone has a good idea, share it. Otherwise each company is forced to spend precious time reinventing the wheel or changing things slightly to avoid lawsuits. In the end, everyone loses.

While writing this piece, it occurred to me that maybe the ultimate user interface should be completely language independent. While this is an interesting concept, current products have evolved to the point that an icon-based design or control surface would have to be quite large to encompass all of the features companies incorporate today. This chart just scratches the surface of all of the commands available. I plan to discuss more advanced editing commands in a future column. ■

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Processing; Sony PCM601 conv, $850; AMS RMX16, $4.5K; 1580, $2.8K; Eventide H3000B, $2.2K; Pultec Mavec, $1.4K; **Mics**; Neumann U67 tube, $2.5K: AKG C12A (tube), $1.4K.

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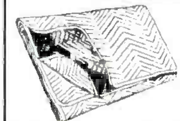

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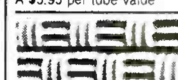

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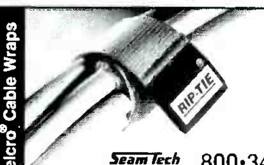

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Audio Visual Assistance	66	41	612-481-9715
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Yamaha International Corp.	19	11	714-522-9011

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