RECORDING ENGINEER/PRODUCER

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August 1985
Volume 16 — Number 4

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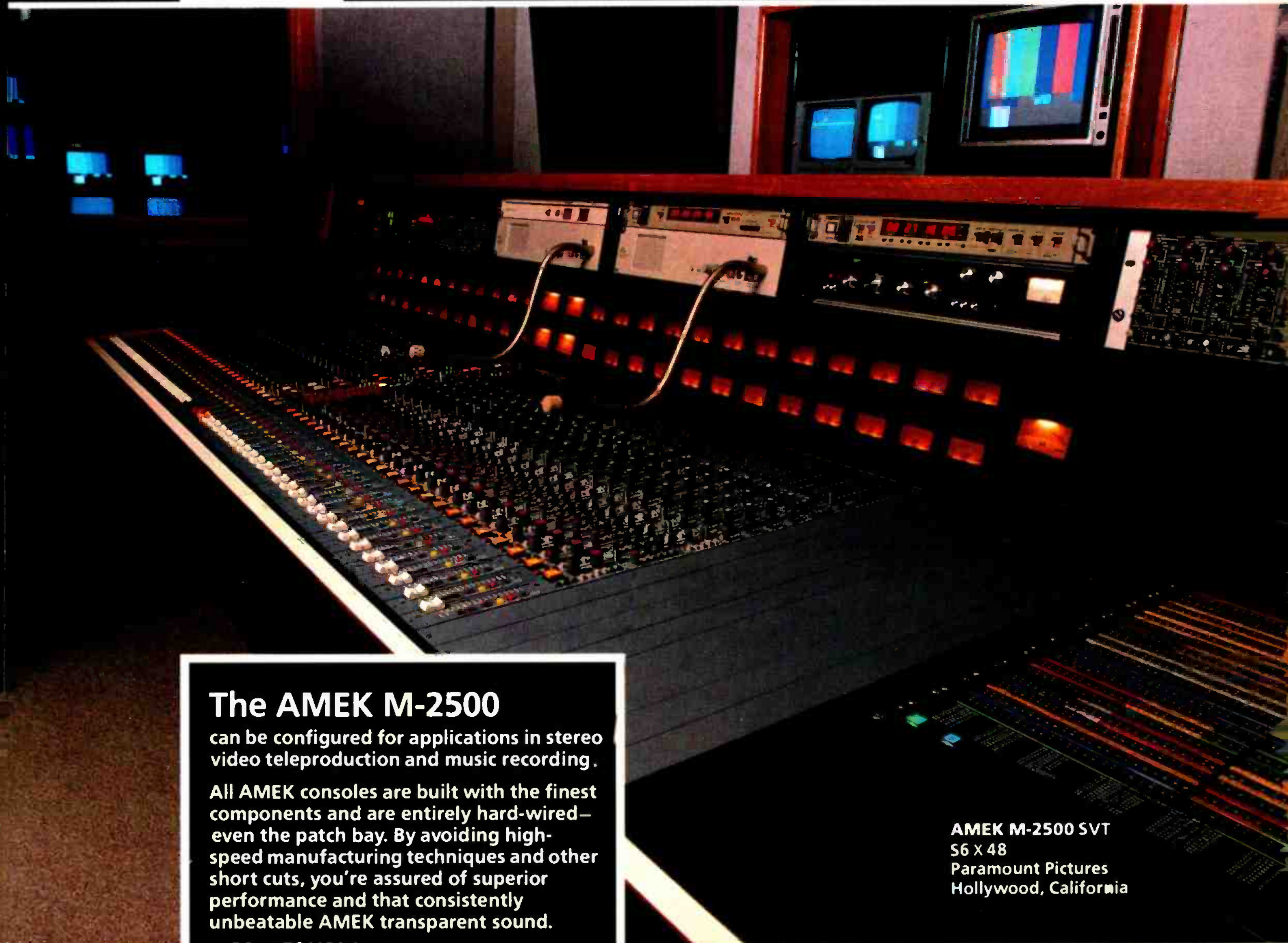

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Paramount Pictures
Hollywood, California

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

— August 1985 Contents —

Volume 16 — Number 4

RECORDING ENGINEER/PRODUCER

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

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

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News Letters Views

DOLBY STEREO ENCODING TECHNOLOGY

from: Gary Reber, VP
Tate Audio
Los Angeles, CA

This letter is to call attention to an omission referenced to the June 1985 article by Larry Blake, entitled "Mixing Techniques for Dolby Stereo Film and Video Releases." The Dolby Stereo matrix decoding circuitry utilized in the Dolby Cat.150 card that converts the two 35mm optical film tracks into four speaker channels, thus completing the encode/decode 4-2-4 process, is the Tate System. Tate Audio, the technology company who developed the full-logic directional enhancement matrix decoding circuitry, licensed its proprietary surround stereo technology to Dolby Laboratories in 1977 for exclusive professional motion picture soundtrack production and theatre exhibition.

The Tate System, as configured in the Dolby Cat.150 card is presently in use in the vast majority of the current 6,200 theatres worldwide equipped with Dolby Cinema Processors. It is the Tate System tchnology which has provided the excellent separation and other performance attributes of matrix decoding in the Dolby Stereo cinema system.

As of last year, Tate Audio modified its position with respect to Dolby Laboratories and, as such, Dolby Laboratories no longer has an exclusive license with Tate Audio. □□□

Larry Blake replies:

I take issue with Mr. Reber's implication that Tate Audio "developed the full-logic directional enhancement matrix decoding circuitry" used in Dolby SVA surround sound techniques used since 1977.

As stated in my February 1981 *R-e/p* article titled *Mixing Dolby Stereo Film Sound*, the first 35mm Dolby Stereo film with surround decoding was *A Star Is Born*, released in late 1976. All of the Dolby Stereo films released in 1977 and 1978 — including *Star Wars, Close Encounters of the Third Kind, Saturday Night Fever, Grease* and *Superman* — were exhibited with the original Cat. 116A/B cinema decoder card employing Sansui QS matrix decoding.

The Cat.150 cinema decoder card using Tate ICs was not in theaters until Spring 1979; again, this was already stated in my 1981 article. As is my understanding, no theater decoder prior to that time contained Tate ICs.

Therefore, I do not understand how Mr. Reber could possibly state that the "Dolby Stereo matrix decoding circuitry . . . [is] . . . the Tate System," since Dolby had licensed stereo motion pictures for over two years prior to their involvement with Tate Audio.

My 1981 article had already outlined the history of the Dolby Stereo, and I thought it more important in June 1985 to concentrate on the practical aspects of mixing for Dolby Stereo release. The only news that I possibly omitted from the new article is that, as stated in Mr. Reber's last sentence, Tate Audio and Dolby Laboratories did indeed recently undergo a change in their business relationship — Dolby is no longer using Tate components in their theater equipment. ■■■

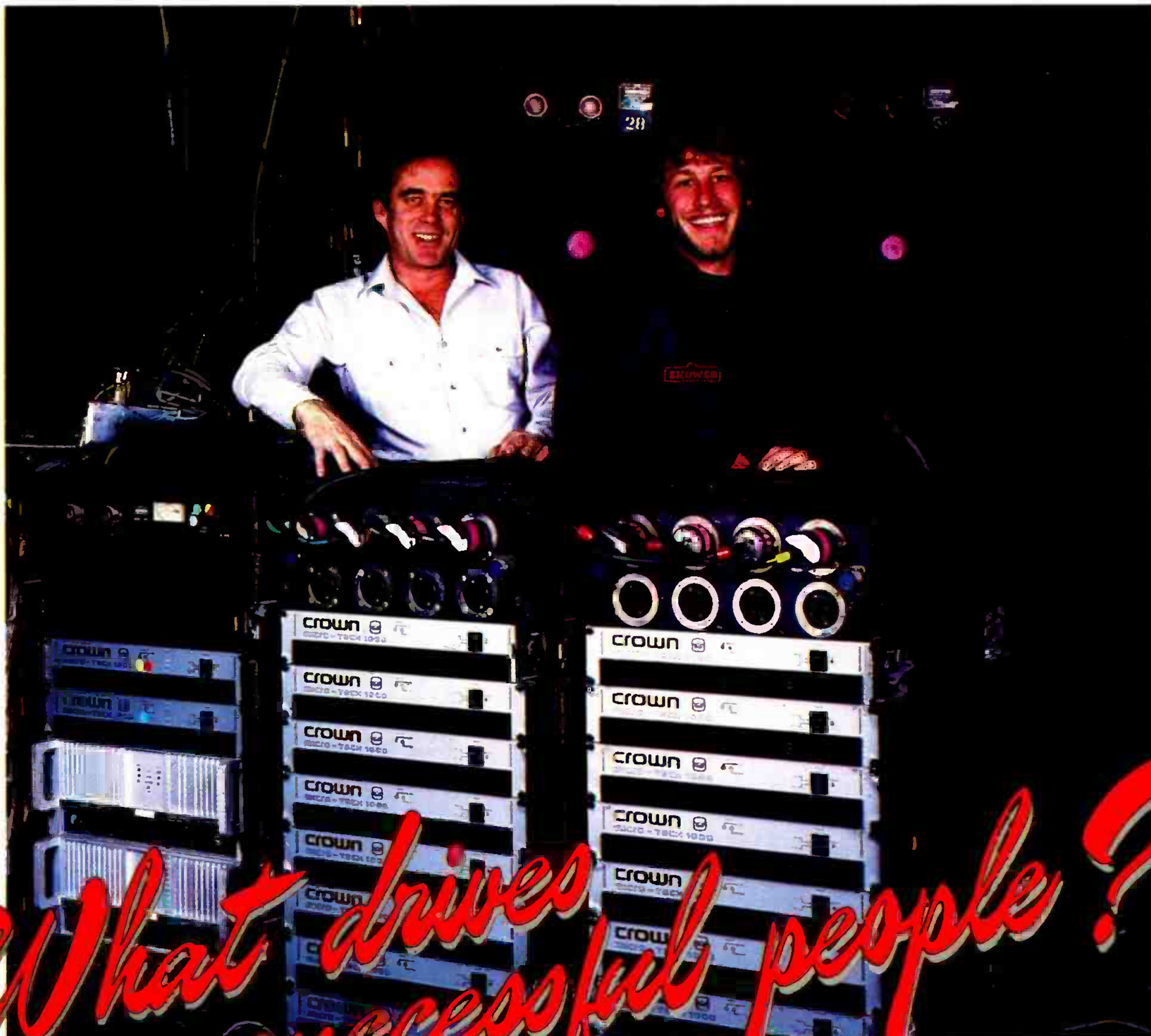
CARE AND REPAIR OF THE STUDIO PATCH BAY

from: Ian Eales, chief engineer,
Garden Rake Music,
and Phil Mendelson,
mx Audio Services

One of the most potentially troublesome items in the studio is the patch bay, but it really needn't be if a little maintenance and common sense is applied. Before we blame the patch bay totally, we should point out that most of the problems we see can be attributed instead to the patch cords themselves.

The air of our cities is not the best, to say the least; a multitude of pollutants are present in the air from automobiles and industrial sources. These react with the metal on the patch cords and jacks to form a tough oxide, which degrades the quality of the signal passing through them. What is required is a method to remove the oxide that has built up on the metal. Any cleaner which does this without leaving behind any waxy residue can be used to clean patch cords. A metal polish such as Mother's Mag Polish does a great job. Take a clean, lint-free cloth, put some cleaner on it, and rub the connector tip. With another clean, lint-free cloth, rub the cleaner off and, presto, you have a shiny, better-than-new connector.

Another quick way to clean patch cords is possible more expensive, but takes much less time and may be more economical if you have a great many to clean. Find a machine shop with a buffing wheel, and ask if you can use it or rent time on it for about a hour. Either use a cleaner like Mother's, or a commercial buffing compound that they probably have at the machine shop. Apply a small amount of compound to the buffing wheel, and then buff the tip of the patch cord. A little practice will



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The performance of the new MT-1000 further strengthens the dependability of Crown amplifiers. The only reason Showco has had to touch their new amps is to move them. And move they have, 78 shows in 52 cities without a failure of any kind. Not surprising for a Crown product but unheard of in any product fresh from the assembly line. Months of Showco's heavy duty field testing has established the MT-1000's reliability even before it hit the market.

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For additional information circle #5

See the first console specifically built for 64 track digital recording at the New York AES

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DESIGNED FOR DIGITAL

Through critical analysis of design, and testing and re-testing of components, the signal path and sound quality of this console is optimized for digital recording. Quad Eight, as a part of the Mitsubishi Pro Audio Group, developed this console as the perfect companion to digital multitracks such as Mitsubishi's new X-850 32-channel recorder.

64 MIXING BUSSES

The SUPERSTAR has 64 mixing busses controlled from a central assign panel and readout. The 72 by 64 output matrix uses logic-controlled summing bus switching, providing 64 instantly selectable output busses. Using its own memory for five complete presets, it also allows automation control via a serial communication port.

COMPUMIX IV AUTOMATION

A 32-bit master processing computer records data on an 80 megabyte Winchester hard disk in real time for unprecedented accuracy in an automation system. This fourth-generation design stores four instantly accessible real time mixes plus eight compressed mixes on the hard disk simultaneously, and transfers compressed mixes to and from floppy disk. A distributed multiprocessing system, Compumix IV has individual computers handling dedicated functions at different levels of the system architecture.

INTELLIGENT DIGITAL FADER

With its own microprocessor, the IDF can operate standing alone or coupled to the automation system. Using a monolithic direct digital 8-bit encoder/fader and a membrane touch panel inputing the 10-bit internal processor, exact dB values are calculated using 14-bit arithmetic, displayed, and converted to DC using a 12-bit D/A. All functions are at 10 times scanning rate for 1/10 frame mute accuracy, and fader smoothing algorithm. There are 16 nested groups, and any module can be assigned master without changing its individual function.

The VCA circuitry is on a separate PC card that plugs onto the main module PC board. Different VCAs may be easily substituted.

PLUG-IN EQUALIZER

Finally, there's a choice! The SUPERSTAR equalizer plugs in on each input module. Normally delivered with a four-band parametric equalizer with variable frequency, bandwidth, and peak/dip level; others are available. Each module also has a variable concentric high pass, low pass filter with individual in/out buttons.

AUTOMATED EQUALIZER

Each channel module has been designed to accept an automated equalizer, making the SUPERSTAR the most advanced console available.

PLUG-IN PREAMPLIFIER

Each module's microphone preamplifier is also of top panel plug-in design. Transformers—or transformerless differential, the choice is yours. And new technology can be instantly added to your console.

Mitsubishi X-850
32-Channel Digital Audio Recorders

For additional information circle #6

tell how much compound to apply. When it starts taking too long to clean a connector, apply more compound. With this method you can clean about 100 patch cords in an hour. When done, check to make sure there isn't any excess cleaner on any of the connectors that could recontaminate the patch bay.

Once these steps have been taken, it is likely that most of the patch bay problems will already have disappeared. If problems are still being encountered, then the patch jacks themselves need attention. The environment in a recording studio is very dry, due to heat produced by the equipment and air conditioning. This makes for an environment prone to generating high static particles which, in turn, attracts small airborne particles of dust into the studio. Invariably, some of this dust settles into the patch bay causing the familiar crackling when a patch is made.

Particles of dust in the patch bay can best be removed with high-pressure compressed air. The patch bay should be removed from its mounting and be freely blasted with the compressed air. If phantom powering is employed on the microphone bay, the voltage supply should be turned off to avoid any danger of shorting the supply to ground while removing the bay. After the dust has been removed, a very small amount of contact cleaner, such as Cramolin R2, may be applied and the bay then blasted again to remove any excess Cramolin. If the bay is particulary dirty, it may be necessary to insert patch cards to open the normals to remove the contaminants caught between the points. It must be stressed that only *small* amounts of Cramolin should be used, and infrequently. Because it is impractical to use Cramolin in bottled form on patch jacks, great care should be taken with the spray, as this often leads to over-use; the use of the R2 (diluted) formulation is acceptable.

One common misconception is that an abrasive is required to remove oxide and contact: the action of inserting the plug wipes contaminants when, in fact, the patch jack with its mating plug is designed as a *self-cleaning* contact: the action of inserting the plug wipes the contact area. Experience shows that burnishing with an abrasive might show an immediate improvement in performance, but only insofar as it has removed the contaminants. Burnishing can compromise the original compatibility of the mating surfaces. Furthermore, continued burnishing eventually removes enough metal that proper contact loading is no longer possible; this is especially true with relay contacts with small contact areas. In the case of contacts that have been plated or flashed with a conductive surface, the consequences of burnishing are obvious. One further consequence of burnishing (or using any foreign metal in a jack), is the depositing of a dissimilar metal onto the original contact surface. This is a potentially serious hindrance to the passage of audio signals, due to the possible creation of a metal-oxide diode.

It should be noted that most patch bay problems are a result of foreign substances. If steps are taken to prevent these contaminants from entering the bay, problems will be kept to a minimum. For example, a simple plexiglass cover over a horizontal patch bay can make a big difference, especially if smokers put their ash trays nearby.

Again, it must be stated that the *patch cord* is responsible for the vast majority of problems. If the cleaning process is repeated every three months or so, you will have better sound for years, and the patch bay will last a very long time.

■■■

Editor's Note: *R-e/p* welcomes short "Hits and Tips" items such as this Technical note from Ian Eales and Phil Mendelson. Send them c/o The Editor to the address given on the Contents page.

"Exposing Audio Mythology" begins on page 15 . . .

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EXPOSING AUDIO MYTHOLOGY

Laying to Rest Some of the Pro-Audio Industry's More Obvious "Old Wives' Tales"

by John H. Roberts

This month we will look into the audible significance of power-supply parameters. In part two, I will move on to consider some more aspects of subjective listening tests.

Just Listen to That Power Supply Will You! . . . Can You?

The majority of respondents to our October 1983 "Golden Ears" questionnaire felt that power supplies *could* audibly affect the sound quality of signal processing gear. Some others have gone as far as to claim the ability to hear different decoupling capacitors in equipment power supplies.

The principle specification used to describe a given circuit's sensitivity to supply interaction (and this month's "vocabulary word") is PSRR, or Power Supply Rejection Ratio. As you might guess from the word "ratio," PSRR is usually specified in decibels; popular op-amps typically specify PSRR in the range of 90 to 100 dB. Discrete circuits can be much better, or much worse, with the simple one-transistor (or one-tube) gain stage having zero dB, or no rejection at all of power-supply interference.

As with other specifications we've discussed in this series, there are subtleties to the correct interpretation of PSRR. To keep this discussion manageable, I would like to limit it to high-performance audio circuits using op-amps.

The first subtlety to proper interpretation of PSRR is understanding that the number commonly quoted only represents performance at a *spot* frequency of say 60 or 120 Hz. As is true of most op-amp characteristics that are improved by negative feedback, the falling open-loop gain required for stability will cause a commensurate performance degradation with increasing frequency. I measured a few popular op-amps at 20 kHz, and found PSRRs in the 30 to 60 dB range. It is also worth noting that op-amps will often have a different PSRR at their plus and minus power-supply rails; one amplifier that I measured exhibited a 20-dB difference.

Another subtlety to using PSRR is understanding that the specified rejection is referenced to the input. Therefore, the op-amp's closed-loop gain must be *subtracted* from the PSRR to reference it to the output signal. For example, an op-amp with 30 dB of PSRR operating at a closed-loop gain of 40 dB will actually *boost* any signal present on the power supply rail 10 dB!

The final piece of information that is important for putting a consideration of PSRR into perspective is: What kind of signal will be present on the power-supply rails under normal operation? I will further break down this power-supply signal down to two types: power-supply induced; and signal induced.

PROFESSIONAL MUSIC ANALYSIS

Readers interested in taking advantage of the services of Professional Music Analysis, a company offering evaluation programs for engineers, producers and musicians intending to break into the music business (see: Jim Riordan's article on PMA published on page 32 of the February issue) can reach PMA at the following address:

Professional Music Analysis, 8761 Katella Avenue, Anaheim, CA 92804. (714) 527-4636. Toll free: (800) 328-8660. Contact: Jan Olson or Mike Wachner.

We did not include address details in the February article because, at the time of its preparation, PMA was planning to relocate to new offices in Anaheim — *Editor.*

Power-supply induced noise would be the ripple in a typical unregulated supply, or the hiss and much lower-level ripple present in a typical regulated supply. The signal-induced component of power-supply noise would be the product of power supply impedance, and changing load current being drawn.

Today, very few high-performance audio circuits use unregulated power supplies, since three-terminal regulators are not very expensive (less than $1). However, the fact the open-loop gain and thus PSRR is maximum near ripple frequency means that a low closed-loop gain circuit could possibly squeak by without one. It is disappointing to note that some consumer phono pre-amps with their RIAA-dictated 60 dB of closed-loop gain at 60 Hz still use unregulated supplies. No matter how well they are shielded, power-supply ripple will appear in the output.

Inexpensive three-terminal regulators attenuate the ripple an additional 70 dB, making it a non-issue for most op-amp circuits. Since such regulators are based on low-performance op-amp technology, their outputs contain notable wideband noise. This noise and reduced, but still present, ripple can be audible in simple discrete or hybrid circuits that exhibit poor PSRR.

Signal-induced power supply noises are a bit more complex to analyze; they are typically induced by the amplifier delivering current to a load. In some cases driving feedback networks can draw significant current. I have seen several analyses incorrectly blame capacitors used in equalizer or frequency-shaping feedback networks for drawing unlimited charging currents under transient conditions. In all cases the maximum current will be defined and limited by a resistance.

As can bee seen from Figure 1A, the current that must be supplied by the op-amp output is a simple function of the input voltage divided by R2. In Figure 1B, the output current will be equal to the input current, which will also equal the input voltage divided by R2. In both cases the current will be completely independent of the value of C1. Where some designers have gotten into trouble is making R2 too small for the op-amp to drive to appropriate levels.

A case where capacitance can and has caused problems is when it is connected

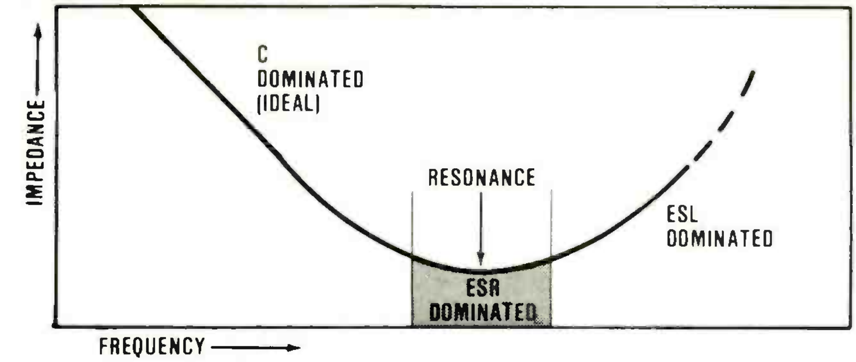
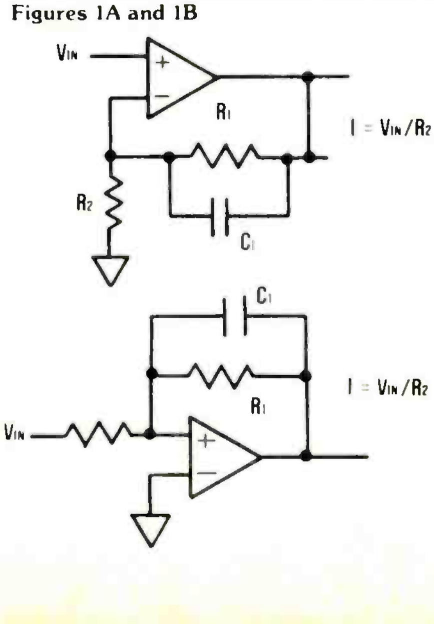


Figure 2: Output impedance of unregulated power supply.

Figures 1A and 1B



between an op-amp's output and ground. No designer would (should?) ever do this on purpose, but connecting even several feet of trusty old shielded wire (at 30 pF per foot) has much the same effect. For this reason it is a good practice to use build-out resistors (50 to 500 ohms) in series with cables to decouple this capacitance from the op-amp.

For the sake of perspective, I should note that power-supply interaction is not the dominant consequence of not decoupling your cables. All op-amps have some internal series-output resistance, which will form an RC with the cable capacitance, delaying (phase shifting) the feedback signal. This phase shift eats up valuable stability phase margin (see my April 1984 column), often causing spurious oscillation. ("Gee Beaver, how come it doesn't oscillate on the bench?") In fact, it is not that uncommon to encounter marginal circuits that break into oscillation when the capacitance of a 'scope probe is connected to the output.

We have already determined that op-amps have a PSRR that deteriorates with rising frequency. Since an op-amp called on to drive cable capacitance will be pulling more current with rising frequency, let's take a look at how the power supply will perform.

The simple unregulated power supply will have a predominantly capacitive output impedance, falling with increasing frequency until reaching resonance and then rising again. Resonance (this month's second "vocabulary word") occurs when the non-ideal resistance and inductance terms (see my February 1984 column) of a capacitor limit its minimum impedance. Beyond resonance, the inductance will dominate for an increasing impedance with rising frequency, as shown in Figure 2.

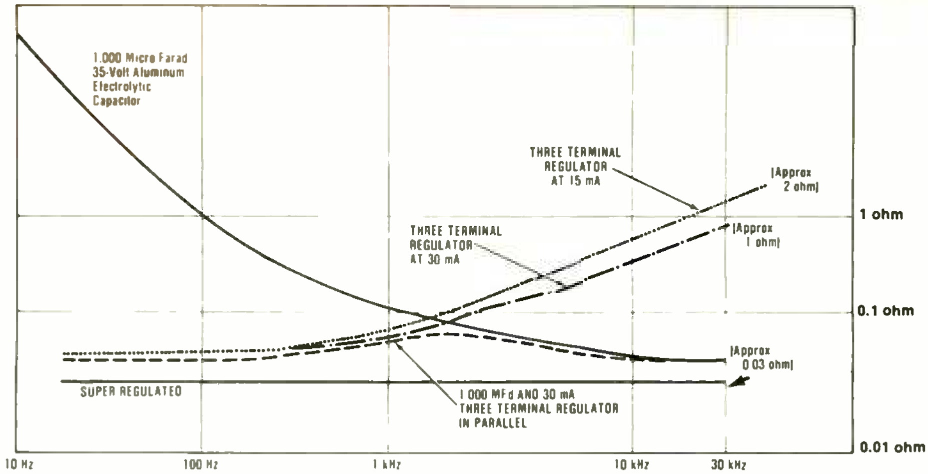


Figure 3: Power supply output impedance variations for single capacitor and three-terminal regulators, separately and in parallel combination.

Many large electrolytic capacitors will reach resonance before 20 kHz, but still have acceptably low impedance over the audio band. (Note: newer capacitors designed for use in HF — 20 to 200 kHz — DC-DC switching regulators have higher resonance frequencies, and lower ESRs).

The output of a regulated supply will have much lower impedance than an unregulated circuit at low frequencies. However, the popular three-terminal regulators rely upon negative feedback to keep their output impedance low. Use of low-performance op-amp technology within causes output impedance to rise at higher frequency, as shown in Figure 3.

It appears that we have a convergence of several unfavorable trends. A capacitive load such as shielded wire will draw increasing current just as the power supply is less willing to supply it. To make matters worse, op-amp PSRR is also falling. Which explains one of the more common causes of interchannel crosstalk, and explains why it is usually much worse at higher frequencies.

A secondary problem caused by rising power-supply impedance is its effect on signal purity (distortion). I had difficulty in reliably measuring distortion caused by inserting even 10-ohm resistors in series with the power leads of a few different op-amps. On the tests I tried, distortion was dominated by the op-amp output stages trying to drive the lower load resistances needed to generate significant voltages at the power rails.

An even more subtle mechanism that I didn't even try to measure (my trusty old test bench only measures distortion products down to -90 dB or so) is the premise that non-linear power-supply impedances or, for that matter, non-linear loads into linear power-supply impedances will generate non-linear signals on the power-supply rails. This is worth considering, since simple Zener shunt-type power supplies or even three-terminal regulators at low current drain will not be particularily linear. While the whole concept of PSRR predicts that these non-linearities will find their way back into the signal, the relative difference between power-supply and load impedance ensures that they will be very small.

Conclusion: It is my opinion that power-supply design typically will *not* affect sound quality. However, as is often the case, poor design practices can always prove the exception. Also, simple topologies, such as tube designs, can be much more susceptible to power supply irregularities.

Recommendations:

- Use high-performance op-amps and, whenever possible, operate them at low (20 dB or less) closed-loop gains.
- Use adequate capacitance in parallel with three-terminal regulators. A 1,000 microFarad capacitor will usually be enough to ensure less than 0.1 ohm impedance across the audio band.
- If you can, band-limit the signal early on. It is easier to roll-off out-of-band-transients, than brute force them out of your power supply once they get in there.

It should be noted that three-terminal regulators exhibit a lower output impedance as they deliver more current — it is a good idea to pull at least 25 or 30 mA from them. For any reader interested in brute-force solutions, see Reference #1 for a power supply designed around high performance op-amps that delivers less than 0.04 ohms across the audio band with very low noise.

Do You Hear What I Hear?

In past columns I have offered various cautions and criticisms of subjective or uncontrolled listening tests. As a circuit designer I learned long ago that things can sound good or bad for all the wrong reasons. The unreliability of such listening tests was recently driven home to be by an experience I had while wearing my other hat as a designer of audio products.

Our brand-new phono pre-amp was reviewed by two different, low-circulation (underground?) audio journals. To quote reviewer #1: "Exciting dynamics . . . suddenly returned. The soundstage widened and deepened, and a wealth of musical detail and energy emerged. Particularly remarkable was the clarity of the midrange and sweetness of the high end . . ."

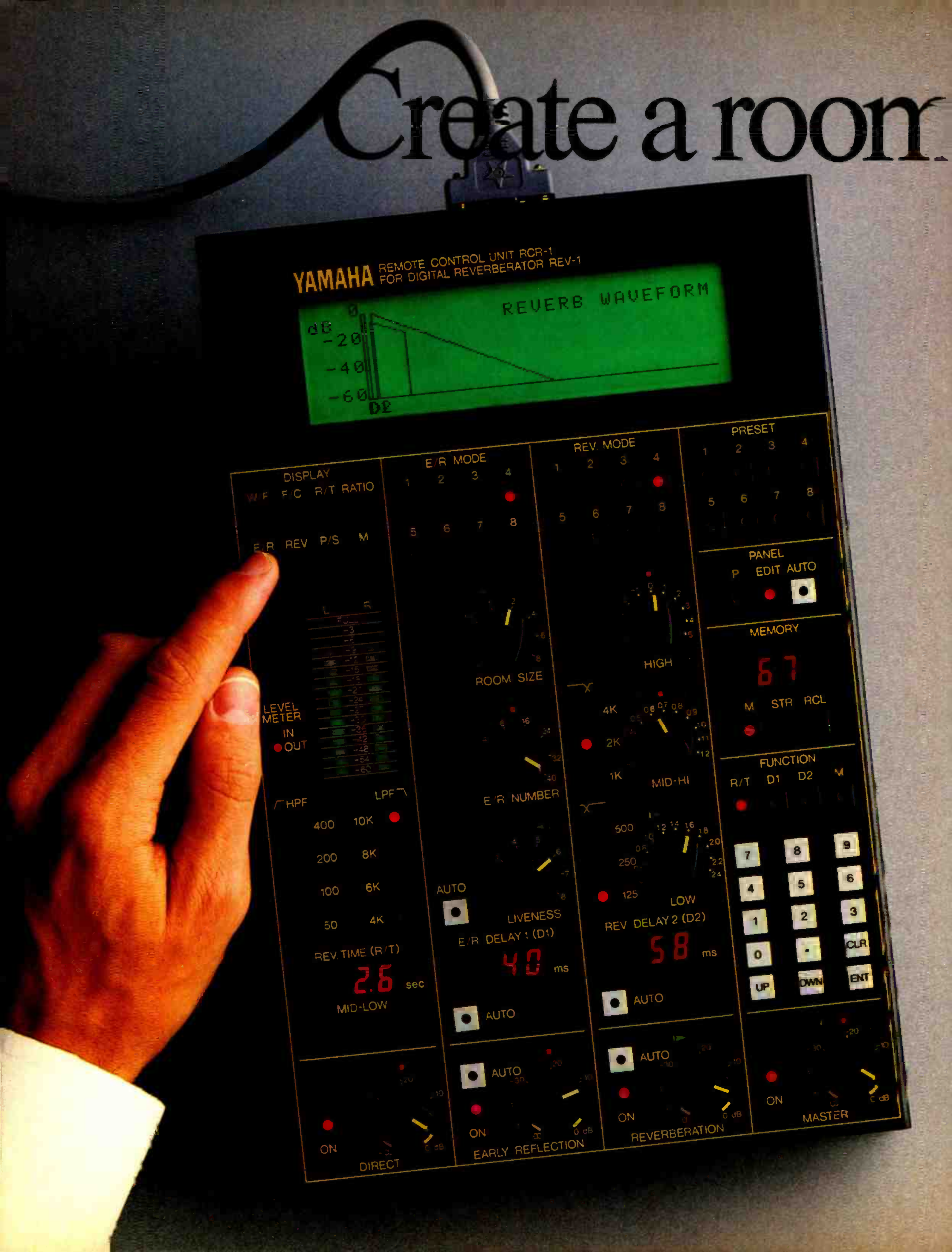
However, the second reviewer had a slightly different opinion: ". . . The ambience sounds . . . were muddled and distorted on the test unit, suggesting indistinctness rather than depth . . . the flute sounded hooded rather than recessed . . . The dryness of the sound made string instruments sound somewhat wiry . . . The voices sounded shallow, without body or support. [It] also lacked the ability to handle complex sounds without mushing them together into a composite sound . . . [and] was unable to follow a second or third voice or instrument with fidelity." And so on.

I personally bench tested the review unit that was the object of so much scorn, and found it working to spec. The subject of the favorable review hasn't found its way back to the factory yet (funny how that works.)

I do not have a good (printable) explanation for how two presumably skilled listeners can listen to two *identical* units, and draw such diametrically opposed conclusions. This would be funny if it wasn't for the fact that some people out there actually read and believe those reviews!

I have presented this example because it is dramatic. Most of us in this business are routinely called upon to make

. . . *concluded on page 23* —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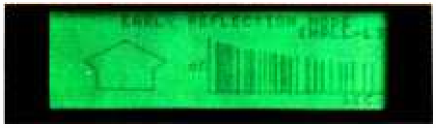
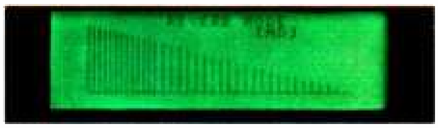
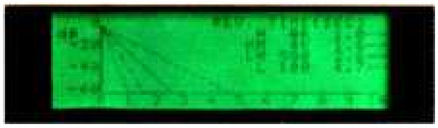
We'd like to open your eyes to the incredible REV-1 digital reverb. Because it gives you unheard-of control over virtually all reverb parameters. And something that has never been seen in any type of reverb: the capability to "look" at the sound as well as hear it.



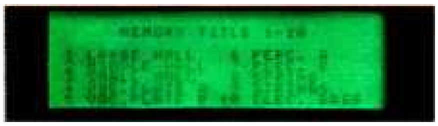
"EARLY REFLECTION" display mode showing room size and relative level and time of discrete reflections.



"REVERB DENSITY" display mode showing level and relative time of subsequent reverberation.



"REVERB TIME" display mode showing difference in reverb time in each of four frequency bands.



"MEMORY TITLE" display showing the titles of internal ROM memories.

The remote unit that controls the nineteen-inch rack-mountable unit has a lighted high-resolution LCD display that graphically depicts the results of the adjustments you make.

So getting just the right reverb sound is no longer a question of trial and error.

The logical grouping of the parameter controls on the remote also makes it easy to create any effect you like. Then store it in any of 60 memories for instant recall.

The remcte also contains 9 additional RAMs so you can store programs and carry them with you to use anywhere there's an REV-1.

And there are 30 additional ROMs with factory preset sounds. Many of which can be completely edited (as can the user-programmed sounds) by using the LEDs to tell you the set value or indicate in which direction to move the control so you can easily and precisely match the value of the originally programmed sound.

And the sound itself is far superior to any other digital reverb. The REV-1 uses specially developed Yamaha LSIs to create up to 40 early reflections and up to 99.9 seconds of subsequent reverberation. So the effect can be as natural (or unnatural) as you want it to be.

We could go on about the REV-1. Tell you about its 44.1 kHz sampling rate that provides a full 18 kHz bandwidth to prevent the natural frequency content of the input signal from being degraded.

How it has a dynamic range of more than 90 dB for the delay circuitry and more than 85 dB for the reverb circuitry.

But why not take a closer look at the REV-1 at your authorized Yamaha Professional Audio Products dealer. Or for a complete brochure, write: Yamaha International Corporation, Professional Products Division, 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 #16

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For additional information circle #17

— continued from page 19 . . .

important decisions based upon what we hear. How confident are you in your chops? When was the last time you had your hearing checked?

Avoid the trap of getting to comfortable with one listening room, or one sound system. The brain has a funny way of compensating for a system's deficiencies and, before you know it, a flatter system than yours may sound wrong to you. Check out the competition. If their room sounds different than yours, check it out and find our which one is wrong — don't automatically assume that *theirs* is wrong.

I expect my reviewer friends have somwhat different sounding reference systems, and very likely different ideas of what a good sound is. If they were both record producers, one of them would be in trouble (maybe both).

Reading for Extra Credit

As usual, I have referenced a few of my earlier columns for more details on capacitor non-ideal performance, and op-amp/feedback circuit's non-ideal parameters.

Reference #1: The power-supply design is actually part of a phono pre-amp design article entitled "Pre-eminent Preamp," published in the #3-85 issue of *The Audio Amateur* magazine. □□□

News

NEVE'S FUTURE LOOKING POSITIVE FOLLOWING RECENT TAKE-OVER OF PARENT COMPANY

In late-June Neve's parent company, Energy Services & Electronics PLC (ESE) was purchased by the Brammer Group, a diverse multinational company with numerous subsidiaries around the world. However, as predicted by various reports that have appeared in the finacial press during the last several months, Brammer subsequently announced that it plans to sell off Neve Electronic Holdings, Ltd., the group of companies that includes Neve Electronics International, Ltd., Rupert Neve, Inc., Rupert Neve of Canada, Inc., and Rupert Neve, GmbH.

Earlier this year, ESE was the subject of an unsuccessful takeover bid by Peek Holdings, which failed to secure the necessary majority of approval from ESE stockholders. Peek also intended to sell control of the Neve Group following purchase of its parent company.

According to Barry Roche, president of Rupert Neve, Inc., "Now that Brammer has confirmed its intention to sell the Neve Group — and the number of companies interested in buying us is substantial, including many that are highly visible in the pro-audio industry — our intention following the sale of the Group will be to secure Neve's leading position within the recording and production studio market. The purchase will further assist Neve in continuing its extensive devlopment plans, and supporting the pro-audio industry with state-of-the-art analog and digital console systems."

At the end of July, The National Sound Archive, London, was scheduled to take delivery of a Digital Signal Processor (DSP) console that will be used to process analog recordings prior to direct digital re-recording. Future plans call for the addition of a custom designed Sound Restoration System to the DSP, which will be used to remove surface noise from vintage vinyl and cylinder recordings.

Meanwhile, work is progressing on the construction of a full-size DSP console scheduled for installation early next year by West Deutsche Rundfunk, the West German state broadcasting organization, at the new Mozart Center in Cologne. Representatives from WDR paid a visit to the Cambridge, England, factory in mid-July to view the DSP at its 50% build-point, and were reported to be extremely impressed with the digital console's topography and operational flexibility. In addition, Neve recently played host to two separate technical delegations from a leading North American broadcast network, and The Turner Network. Visits to CTS Studios, London, were also arranged, to allow the delegations to gain hands-on experience with the facility's 48-input/32-group DSP console.

Besides success in the digital-console market, sales of Neve analog consoles

News

and automation systems are reported to be very healthy. In excess of $1.4 million worth of NECAM 96 servo-controlled fader automation systems have been sold in the first six months of this year, Roche says.

"The future of the company's activities in North America is extremely buoyant," Roche added. "Merging with a company that has our type of products in its mainstream of business cannot but help to increase Neve's presence in the pro-audio marketplace. We expect to see continuing and substantial growth in the future."

At press time there was no final word on the purchaser of the Neve Group.

SOUNDCRAFT WINS SEVERAL CONTRACTS FROM BBC

Soundcraft Electronics Limited has secured three separate orders from the British Broadcasting Corporation, including a contract to supply the BBC with four consoles that will be used for television applications during the 1986 Commonwealth Games, to be held in Edinburgh, Scotland, next July. The four customized Series 500 consoles will be installed at the purpose-built Broadcast Centre in Edinburgh — currently under construction — for use by both the BBC, as host broadcaster, and ABC Australia.

In addition, the BBC's Bristol Network Production Centre has ordered a customized Series 500 that will be installed at St. George's Church, Bristol for both live broadcasts and recording work.

Finally, the company recently supplied eight, 16-channel Series 200 portable consoles — including flight cases — for use by BBC Radio outside broadcast units in London and throughout England.

SOLID STATE LOGIC OPENS FAR EAST OFFICE

According to SSL founder and managing director, Colin Sanders, "Our studio and broadcast systems have become very popular in this part of the world, and we forecast considerable additional growth throughout the region. SSL Far East will provide these clients with the kind of technical support and information exchange enjoyed by SSL owners and users in the world's major production centers."

The new Hong Kong office will serve SSL clients in China, Singapore, the Philippines, Malaysia, Indonesia and Thailand. Currently six staff members are based at the new location — all of whom are multi-lingual — operating under the direction of Solid State Logic Far East's managing director, Bingo Tso.

Tso, a former Hong Kong-based record producer for Polydor, EMI and WEA, and who has served as an SSL consultant in the region for the last two years, explains that a primary reason for opening the office is the emerging Chinese recording industry. Three SSL SL4000 E-Series Master Studio Systems were installed at the China Record Company recording studios in Beijing, Guangzhou and Shanghai last year, and a fourth SL4000 console is currently on order by CRC.

Solid State Logic Far East is headquartered at Austin Tower, Suite 301, 22 Austin Avenue, Tsimshatsui, Kowloon, Hong Kong. Telephone: (852-3) 721-2162; FAX: (852-3) 723-5465; Telex: 47580 (SSLFE).

CIRCUIT RESEARCH LABS ACQUIRES ASSETS OF MICMIX

Circuit Research Labs, Inc. has purchased certain assets of MicMix Audio Products, Inc. These assets consist of inventory, patents, licenses, contract rights and business name.

According to Bernard M. Van Benthem, president of CRLI, "This acquisition marks our entry into the professional recording marketplace, an important new market for the Company. Furthermore, we are confident we will be able to develop and market additional new products based upon MicMix's patented audio devices, thus placing us in a much stronger competitive position overall."

For additional information circle #20

Circuit Research manufactures and markets electronic audio-processing and signal-generating equipment used to enhance the quality and range of broadcast signals transmitted by radio and television stations in the U.S. and abroad.

GOTHAM AUDIO CORPORATION PURCHASED BY GROUP HEADED BY RUSSELL O. HAMM

Stephen F. Temmer, company founder and president has announced the sale of Gotham's business and name to a group of investors headed by Russell O. Hamm, long-time VP. The transaction, which involved an undisclosed amount of money, was effective April 1, 1985.

Temmer will retain ownership of the Gotham Building in New York; Quantum Audio Labs, Inc.; a substantial interest in Gotham A.G., of Regensdorf, Switzerland; and Thorens-Franz AG of Wettingen, Switzerland, manufacturers of Thorens turntables.

Russell O. Hamm, president of the new company, which retains the Gotham name, has announced several immediate changes to bolster the company's relationship with customers and suppliers. While product lines from Neumann, EMT, NTP and Klein & Hummel will continue to be the foundation of Gotham's business, Hamm states that he will step up sales and promotional activities in an effort to reach a broader customer base. "We plan to maintain a higher profile in the future, not only making more people aware of the high quality products coming out of Europe but particularly to make them aware that these products are affordable."

Jerry Graham has returned to Gotham as eastern sales manager after an absence of several years, and has been working with Hamm on a new EMT dealer program. Juergen Wahl, Gotham's newly appointed west coast sales manager, has just returned from an intensive training seminar with Neumann engineers in Berlin. Juergen has also re-opened the Los Angeles office, which was closed when Hugh Allen, Jr. retired more than a year ago.

— NEWS NOTES —

• Nine pro-audio manufacturers have donated the use of nearly $200,000 in equipment to the **ASSOCIATION FOR MULTI-IMAGE INTERNATIONAL (AMI)** to provide sound for its festival to be held in Dallas, Texas, August 12 thru 17, 1985. The contributing companies include **OTARI CORPORATION, QSC AUDIO PRODUCTS, APHEX, COMMUNITY LIGHT & SOUND, PRO CO, SOUNDCRAFT ELECTRONICS, WHITE INSTRUMENTS, AUDIO-TECHNICA U.S.** and **TECHNICAL PROJECTS**. This year's festival marks the fifth year of an on-going effort to raise the audio sights of multi-image producers and staging companies. In addition to the equip-

News

ment donations, the **SOCIETY OF PROFESSIONAL RECORDING STUDIOS (SPARS)** will conduct two seminars concerning planning, budgeting and producing original music tracks for multi-image shows. According to Emory Straus of White Instruments, great improvements have been made in the production and reproduction quality of multi-image audio since the project began four years ago.

• **KBA ENTERPRISES** and **FLANNERS PRO AUDIO** has announced the 1985 **CHICAGO MUSIC EXPO**, to be held at the Hotel Continental, Chicago, September 20 thru 22. Planned as the largest consumer music show in the Midwest, Expo '85 will feature exhibits, talent showcases, seminars and an industry awards banquet. The exhibits, sponsored by Flanners, will feature the latest in pro-audio equipment and musical instruments, all of which will be available to the pulic for sale at the show. Expo organizers say that currently there is no other show that allows manufacturers to "come into such direct, positive contact with their customers." The seminar program comprises panels of regional and national experts that will address such subjects as Producing Music Videos, Direct Marketing Techniques for Independent Records, Music for Jingles, and Recording Studio Management. Further details of Expo '85 are available from KBA Enterprises, Inc. 110 Schiller Street, Elmhurst, IL 60126. (312) 279-8323.

• **AUDIO INTERVISUAL DESIGN**, the LA-based pro-audio equipment systems company, has been appointed Southern California dealer for the new Model 833 Studio Reference Monitor from **MEYER SOUND LABORATORIES**. According to AID president, Rick Plushner, "The design [of the 833] is perfect for professional studios, especially in close-field monitoring applications, which also makes it the ideal speaker in the smaller, privately-owned studio. The Meyer speaker produces maximum linearity at both high and low levels, and has a degree of accuracy and realism that is phenomenal." In addition to the 833, AID is marketing the Meyer Model 834 subwoofer and CP-10 Complementary Phase stereo 10-band parametric system equalizer.

STOP PRESS:
ATC to Distribute Telefunken Products in North America

Ranier Zopfy, sales manager of AEG Telefunken Corporation, has announced that ATC will import and distribute the Telefunken line of tape machines and products throughout North America.

"AEG has made a long-term commitment to this market," Zopfy says. "We have developed audio recorders designed specifically to meet the needs of the market, and have maintained our own personal commitment to manufacturing only the finest equipment that German engineering can design. We have already committed to the large inventory and technicians necessary to support a market as large as the United States. We will be naming our East and West Coast representatives shortly, and establishing regional inventories and service. By importing and distributing our own equipment, we are now very competitively priced."

The new company address is: AEG Telefunken Corporation, Route 22- Oor Drive, Sommerville, NJ 08876; (201) 722-9800.

• To expand system production to meet increased demand, **EASTERN ACOUSTIC WORKS** has added 2,500 square feet of assembly space at its Framingham, MA, location. Along with the increase in space, EAW has remodelled all assembly facilities to create separate production areas for small systems — MS30 thru FR-153 — and larger systems — FR22 thru KF550. In August, the company's Customer Service will become a separate department with new personnel to improve turn-around time on repairs.

• **THE WESTWOOD ONE RADIO NETWORKS** has acquired **STARFLEET COMMUNICATIONS**, one of the nation's leading producers of live radio concert programming. Following the acquisition, Sam Kooper, founder and president of Starfleet, and now serving as Westwood One's director of

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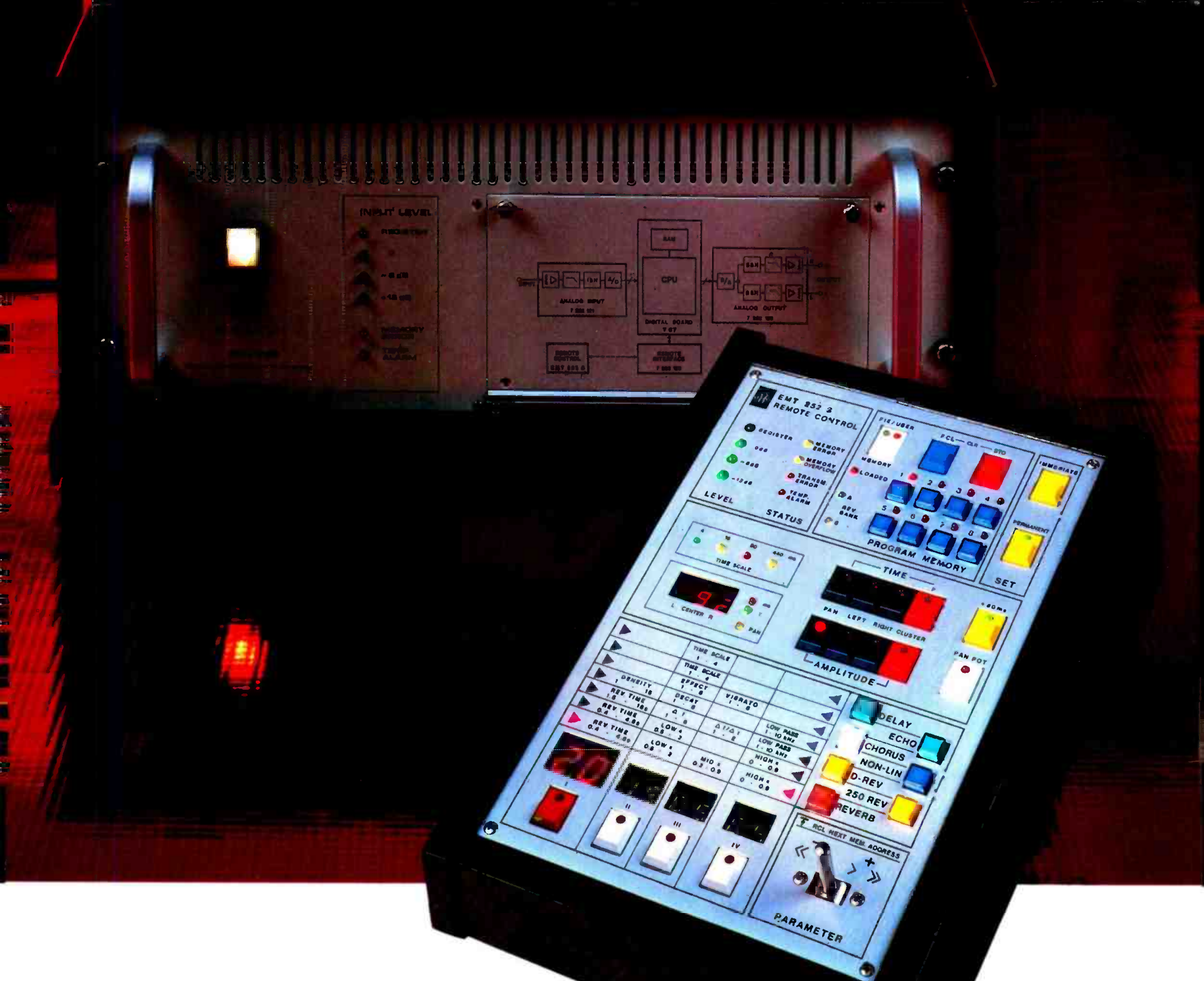

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Our very first unit, the 140, is *still* used by major studios. And the sound of our big digital units, the 250 and 251, is legendary.

Now EMT introduces their new concept of digital reverb—the EMT 252. It offers *more* programs and features than the 250 and 251. Here's what we mean. . .

- ☐ Functionally clear, easy to operate control panel.
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- ☐ 40nS digital processor speed coupled with 16 bit A/D converters yields 15kHz bandwidth and full dynamic range.
- ☐ The original EMT 250 reverb program for superb sound at short decay settings.
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- ☐ Chorusing, delay, echo loops, doppler reverb, and non-linear reverb programs.

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For additional information circle #69

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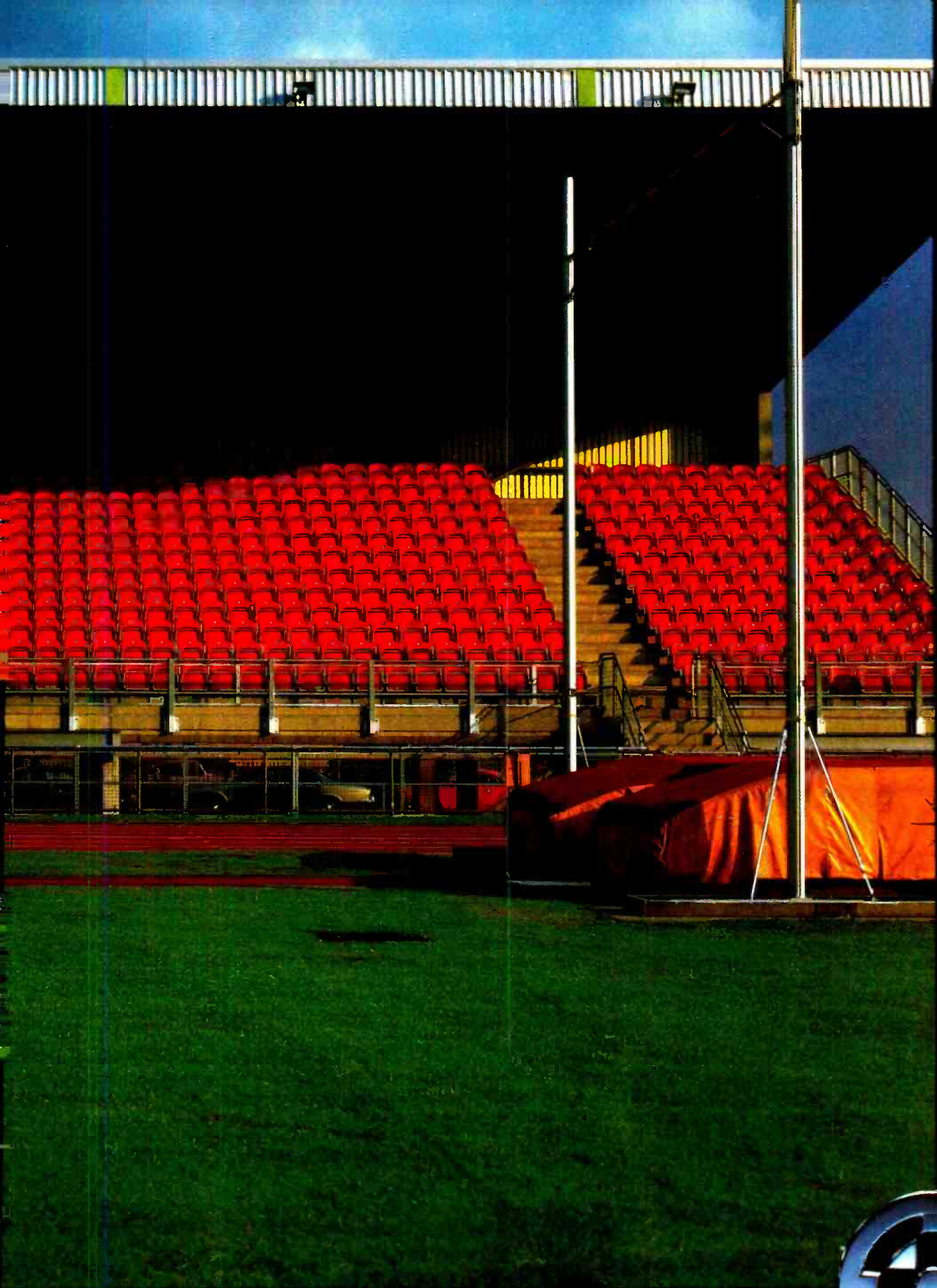
...and some people start a session
without a Klark Teknik Reverb.



KLARK TEKNIK

Few engineer/producers are fortunate enough to experience the kind of meteoric rise that has marked the career of **Daniel Lazerus**. Originally a drummer and keyboard player for a number of small bands, the native Californian drifted to the other side of the glass in 1977. During five years as a second engineer — first at the Wally Heider Hollywood facility, then at The Village Recorder — Lazerus got his training on album projects for Stevie Wonder (*Musiquarium, Part I*), Tom Waits (*Foreign Affair*), the Rolling Stones (*Emotional Rescue*), and Fleetwood Mac (*Fleetwood Mac Live*), as well as two soundtrack albums (*The Wiz* and *Grease*). The big break came, however, when Lazerus was seconding for Roger Nichols on the *Eye to Eye* album project with producer Gary Katz: when Nichols' schedule forced him to leave in mid-project, Lazerus stepped in to take over as principal recordist, and complete the album. Subsequently hired to record Donald Fagen's *The Nightfly*, Lazerus promptly earned himself a 1983 Grammy nomination as Best Engineer for his work on the Fagen project.

As one might expect, the succeeding three years have been very busy for Daniel Lazerus. His engineering credits now include projects for John Denver (*Seasons of the Heart* and *Dreamland Express*), Diana Ross (*Ross*), Jean-Michel Jarre (*Zoolook*), Eddie Murphy's debut musical album (*How Could It Be?*), and "The Finer Things" from the *King of Comedy* soundtrack album. His production credits span work with Joe Cocker (*Civilized Man*), the original cast album of *The Gospel At Colonus*, and "5 Minutes/Bonzo Goes to Washington" (a rap/street dance single with Jerry Harrison of the Talking Heads).

When we caught up with the busy young engineer, he was tracking basics with producer Gary Katz at LA's The Village Recorder for an album project with Rosie Vela.

Daniel Lazerus

Interviewed by Ralph Jones

Photography by Elizabeth Annas

R-e/p (Ralph Jones): In looking at your biography, I'm struck by the variety of artists you've worked with — from Joe Cocker to Diana Ross, John Denver to Donald Fagen. I assume that such a broad range of music would imply a similarly broad range of engineering and production styles. How do you accommodate such differences?

Daniel Lazerus: Basically, it just comes down to relating to the individual artist — genuinely understanding what they really desire. Take, for instance, an artist like John Denver: he could be recorded and produced with drum machines, MIDI-interfaced keyboards, and so on, but it wouldn't be what he's about. By the same token, you cannot record him in the way that he's been recorded before, because that's not what people are listening to these days. So, what's needed is a creative and sensitive balance, and it often comes from an engineering standpoint: how you can "stretch" the sounds. The album that I did with John [*Dreamland Express*], which really is marvellous, has a lot of interesting sounds: some strong and unusual drums, and interesting guitar parts.

It's about balance. You can only go so far — you can't be obvious, but you need to be new and strong. One thing that helps me to achieve that balance is that I'm extremely inquisitive and experimental. When I'm handling a basic tracking date, I'll try and use a different miking configuration than I've used before — if only to continue to test myself and be sure that I'm not just relying on something that has worked before. Both with other producers, and when I'm working in a production capacity myself, I've been able to experiment and yet maintain a sense of the artist's integrity — of course, that's the ultimate aspect of any production.

R-e/p (Ralph Jones): There must be a set of basic principles that guide your

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The Celebrity Series feature large diameter diaphragm/voice coil structures for increased sensitivity with the ability to handle high sound pressure levels. These higher output levels allow for significantly less mixer gain and are a tremendous aid in maintaining good signal-to-noise ratios.

Perhaps the most important characteristic of any performing microphone is reliability. The design of our cartridge/shock mount system increases ruggedness as well as isolation capability to insure long-term performance under severe field conditions.

Our microphone screen utilizes extremely heavy gauge wire that has been "junction locked". Once the screen is formed, we do not stop there. The heavy wire screen is "fired" in an oven after forming, thus causing the plated wire to "fuse" at all interconnecting points. The result is an unbelievably durable "brazed" wire windscreen that will hold together under the most severe abuse. After the ball windscreen is formed, brazed and coated, a precision urethane foam pop filter is fitted to minimize the undesirable proximity effects. This special acoustically transparent foam protects the entire sound system by breaking up explosive high SPL pressure waves created by close vocals or close miking percussion instruments. For those applications requiring even more acoustic screen from wind noise, etc., Peavey offers special external colored wind noise filters that slip over the screen and internal pop filter.

While outwardly, the appearance of the Celebrity Series is somewhat conventional, the aspect of "feel" has been given heavy emphasis since our experience has shown that performers prefer a unit that not only sounds right and looks right, but must also have a comfortable balance, weight, and overall tactile characteristics.

Special "humbucking" coils (models CD-30™ & HD-40™) have been designed into the microphone element that effectively counterbalance any hum that might be picked up from external sources. Performers who play clubs where hum from light dimmer switches or other sources are a problem can appreciate this unique feature.

We invite comparison of our Celebrity Series with other cardioid microphones. You'll see why we feel that in terms of performance, features, and price, there is no competition.

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For a complete catalog featuring the entire line of Peavey sound reinforcement equipment send $1.00 to Peavey Electronics, Dept. A, 711 A Street, Meridian, MS 39301

For additional information circle #23

Daniel Lazerus

experiments, besides your own desire to try out something new. After all, the technique has to be appropriate to the material that you're recording. Are you guided in any sense by the artist's past work?

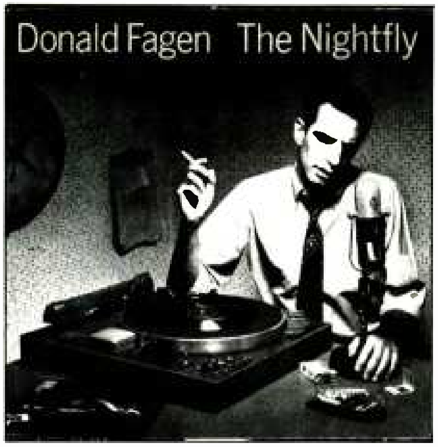
Daniel Lazerus: For me, it's an extremely instinctive process: although I'm certainly aware of the artist's recording history, I don't actually think about it a lot, because what's important is not the past, but the moment in which we're working. Usually, I'll spend time listening to demos before we begin tracking basics, and really get a sense of the project. Then, I spend a lot of time diagramming the miking concepts that I'd like to try: basically, I have those things set up beforehand.

R-e/p (Ralph Jones): Do you tend to be influenced by the sonics of the demo, or do you listen more for song values, and then try to create new sonics to enhance those values?

Daniel Lazerus: I definitely would like to create new sonics. But, these days, demos are so sophisticated that they often can serve as an incredibly accurate guideline to the way that a song is to be recorded. I'm even impressed by the quality of Portastudio song demos that I hear these days — really, I'm serious — recorded, for example, by someone who lives in a one-room apartment in Manhattan, plays all the parts, and also sings.

One musician like that is Joy Askew, a very different and talented keyboard player who works with Laurie Anderson. Joy gave me some amazing demos, made when she was living in a very small space. She had a mike mounted in a bookcase, and she would walk back into the kitchen to sing, so that she got this kind of "echo-y" kitchen sound. In general, there hasn't been a lot of rearranging involved when we work from demos, but certainly I work to make the sonics much more sophisticated. Even if it's a very live, raw sound that we're after, I try to stretch that to make it *more* live, *more* raw. After all, the artist put that demo together to represent what they heard. It's their song — it's what they want. If you can expand on that, then that's a good thing to try for.

R-e/p: Given your emphasis on experimentation, I would bet that you sometimes are frustrated by budgetary limitations. After all, recording time in the studios that you've been working in isn't all that cheap!



DL: Sometimes it's a bit of a fight: usually, you can only get so experimental in the time you're given. Even so, I try to experiment as much as I can. Jean Michel Jarre's new album, *Zoolook*, was a great opportunity to be experimental, just because of the nature of his artistry. I used things like [Crown] PZM mikes on glass baffles that were about eight feet high and four feet wide. Laurie Anderson performed on the album, and I recorded her inbetween three baffles with PZMs on each baffle — it really worked out great. I used the same technique to record Yogi Horton, who played drums, and was able to capture some very "microscopic" drumming sounds. The toms, particularly, worked out very well with these glass baffles around them holding PZMs.

R-e/p: Your biography says that you "designed" the New York sessions for Jean-Michel Jarre's Zoolook *album. That term intrigues me. What does it mean?*

DL: It is a very juicy term, isn't it? "Designed" — it sounds like inventing tofutti or something! Well, Jean-Michel had never done a studio album with musicians: he's primarily an instrumentalist. He has had huge, wonderful-sounding albums with widespread success in Europe and elsewhere, but everything has been keyboardoriented. He was a fan of *The Nightfly*, which won the Grand Prix du Disque in France — the top album award — for that year. So he looked me up when I was in New York, and we met and talked about the project, which was going to utilize real musicians — not just his machines. It was quite a risk for him. "Designed" simply means that here was someone in France who had some ideas; I was in the U.S., and I helped him realize them. But I also had some ideas of my own: I thought of using Adrian Belew and Yogi Horton. I put the session together while he was in France, and it really fell together beautifully.

R-e/p: You've worked in a number of studios, both here in L.A. and in New York. Presumably, each studio has different characteristics — a different room, console, monitors, tape machines. How do you cope with the differences among facilities, and still maintain some kind of continuity in your work?

DL: That's a difficulty that every independent engineer faces. It's a matter of knowing what you want, and being able to understand a room quickly. Most engineers grew up seconding, and getting to know certain rooms. It's a big step when you begin engineering on your own and start working in a lot of different studios, yet still want to have a central kind of sound — a "heartbeat."

I think that I've been able to keep a "center" about most of the records that I've done, even while working in very different studios. It isn't always easy, and sometimes I'm not so sure that there hasn't been something lost in trying to keep that center. Maybe I didn't know how to simply cut loose, because I was driving towards a sound that *I* wanted, and didn't always understand what the *room* was offering.

It's certainly much more comfortable to have a room that you know and love. But if you are in a situation where you've agreed to work on a particular project in a particular studio, there's really no excuse for complaining: it's a matter of seeing what the room is and dealing with it, even if it's just a shoebox. Obviously, the optimum situation for any engineer or engineer/producer would be a room that they know to be consistent; one that they can experiment with and stretch themselves a little further. It's difficult to go into radically different rooms, but it's also a real test of your capabilities, and I enjoy the challenge. And there's a way to have continuity, also.

R-e/p: What do you look for when you first go into a new studio?

DL: Initially, I think about how the drums will sound. Drums are your foundation: that's really the *basis* on which everything's going to be structured. If a room seems inappropriate for drums, I won't cut them there.

R-e/p: What do you look for to get a good drum sound?

DL: An ability to have a strong, close,

If it's a very live, raw sound that we're after, I try to stretch that to make it *more* live, *more* raw. After all, the artists put that demo together to represent what they heard. It's their song — it's what they want. If you can expand on that, then that's a good thing to try for.

ANNOTATED DISCOGRAPHY

An Inside View to Track Production: A Conversation with Engineer Daniel Lazerus

By way of an introduction, Daniel Lazerus played for *R-e/p* a variety of recordings, drawn from past and recent projects, that he considered representative of his best mixing and engineering techniques. During the listening session, Lazerus offered comments on aspects of each selection:

John Denver — title track from *Dreamland Express* album.

"This may give you an idea of the change that we perpetrated, compared to John's previous recordings. We did this album in a very short period of time: we cut 16 tracks in three days, and mixed very quickly. Even so, we spent some time dealing with the kick drum and snare on this cut, because they were somewhat difficult to record. The drums were played with brushes, and the kick drum was played very lightly to draw out a semi-contemporary sound.

"I miked the kick with a Sennheiser MD-421 placed about midway inside the body of the drum, and added some damping on the head. Then I processed it through a couple of stages of API EQ, and I used a dbx limiter to keep it consistent — because it was just being tapped. Quite honestly, it was a real obstacle, because it was being hit so lightly: I had to work to get it to be linear, and balanced in the track.

"The compression helped to draw it into the line. Then, using a couple stages of good quality, older EQ, I lost quite a lot in the region around 350 Hz; that makes an interesting kind of 'black hole' in the kick sound, though it's still got the lower-end. I also like to add two to four dB at 50 or 60 Hz: I like to have that 'ultra-low-end' cranked a bit, since I know that they're going to roll off some in the mastering, anyway. Of course, EQ won't do much if you don't have the proper drum set or a good mike position. So, during the recording session, I was constantly running out and adjusting the mike an inch to the right or left from where it was, tweaking the positioning to get the best sound.

"The snare was played with a brush, and miked with a Shure SM57 going through a couple of different EQ stages — an API and a Pultec — which I used to draw out the qualities that seemed best for the track. What I'll say about the snare here is that it lacks as much of the hi-hat as possible, while still retaining a good snare sound. I really count on proper miking technique to isolate the snare: I *detest* gating tracks, because I hate to commit to gating in the tracking stage, and then have to deal with it in the final mix."

John Denver: "Gimme Your Love Forever" from *Dreamland Express*

"This album was an interesting challenge for me, because we were working on an old console, and had no computer assistance; all these mixes were entirely manual. This song, in particular, was challenging because it is a fairly complex arrangement: the band was supplemented with a full horn section, with Paulinho da Costa on percussion. Initially, Paulinho played only the conga track; then I thought that it might be great to have timbales and a whistle — because the song has a Brazilian feel to it — so he added those parts in the overdub stage."

We were particularly struck by a background vocal breakdown that occurs at the end of the tune. How did that come about, we queried?

"That was actually my idea. It seemed that, at that point in the song, there was an awkward feel to the timing and phrasing of the track. So I decided that, in order to improve

Daniel Lazerus at the SSL 4000 console, Amigo Studios, North Hollywood.

tight drum sound, and a live sound as well.

R-e/p: How do you tell whether a room is capable of delivering that kind of live sound?

DL: It usually seems to depend on the size of the room, although I've been surprised — I've been in situations where I've been able to get strong drum sounds from pretty small, boxey rooms. But I tend to like a larger room. In the Village Recorder [Los Angeles], for example, Studio D has a walk-in echo chamber. It's a great room that was developed during the recording of the Fleetwood Mac *Tusk* album. To be honest, I'm not certain why it works or what the design is, but it's an *incredible* live chamber that you can set drums in front of, and do some interesting kinds of miking. A very large room like that allows the opportunity to do a lot with basic drum tracks. It allows you to cut the basic drums strong — I mean, those have to be pure — but also to think of recording live-sounding tracks.

R-e/p: Another thing that must be a concern when moving from one studio to another is the whole question of different monitoring systems and "house curves": the interaction between the monitors and the control room. How do you maintain any degree of trust in the result, when these aspects can vary so widely?

DL: When I did *The Nightfly* with Donald Fagen, I became familiar with David Visonik 9000 speakers. Donald, Walter [Becker] and Gary Katz had been mixing on the 9000s, and I really fell in love with the speakers. For every single album, and every studio in which I've worked, I've taken those speakers with me. They're marvellous monitors: they're what *Aja*, *Goucho*, and *The Nightfly* were mixed on. The 9000s have a bump around 60 cycles but, once you understand that, they're "home"; you're taking "home" with you wherever you go, and I think that's really important. I think the Visoniks are a fabulous "heartbeat" — a great center monitoring system.

R-e/p: This "bump" at 60 — do you find yourself tending to compensate in some way for that, or do you just take them as being representative of an average "real-world" consumer speaker?

DL: That's a good question [pauses reflectively]. The bump at 60 is just something that I understand. I deal with it by knowing the speakers; they're just very familiar to me. It's not about compensating: it's just about *good* recording. What we're talking about is a "masking" kind of effect: you don't hear the bass in the area around 60 to 100

ANNOTATED DISCOGRAPHY — *continued . . .*

it, we might try to do an odd vocal breakdown with percussion.

"First, I set up a tape slap that was musically timed with the meter of the hi-hat. I panned the background vocals left and right with an echo that made them three-dimensional, put the tape slap down the center, and also sent the tape slap return to echo. At the breakdown, I dropped most of the tracks with a manual mute; the effect serves to smooth it out so that it's not so extreme. It seemed to jazz up the track and, in the vein that we're talking about, it's something appropriately new for John. A breakdown in a John Denver song is an unusual idea, and doing it with the background vocals seemed like an interesting way to go about it!"

Rosie Vela: "Zazu" from untitled debut album, projected for release in late 1985.

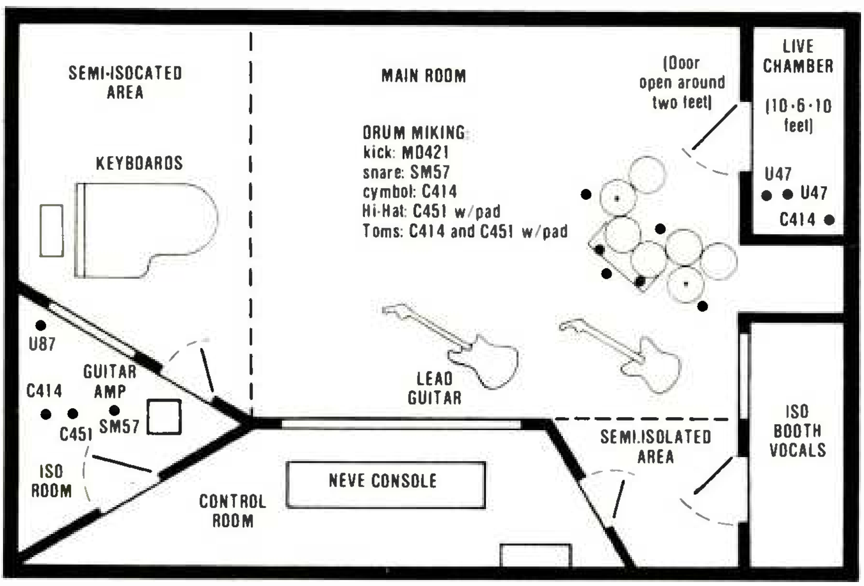
"Rosie is a model, probably one of the top ten on the Planet Earth. She has written a lot of marvellous songs that are sort of eccentric — a little like a Kate Bush or a Peter Gabriel. I really believe this is going to be a strong record; certainly I can suggest that she's written some very interesting songs. I think she'll have a real strong possibility as an arriving new artist.

"I'm particularly proud of the drum sound on this cut. The kick is very strong and deep, and it hits you, but it's not boomy. It's a matter of using enough of the actual kick sound — which was a strong, bottomy, straight-ahead kick drum — and mixing it into a live acoustic chamber [at the Village Recorder, Los Angeles]. To get the chamber sound, I was rolling mikes around the whole time, to everybody's chagrin!

"I used Neumann U87 mikes in the chamber, and they were highly compressed, so they're 'sucking' in that room — which also helps to 'snap up' the delay time that can occur. The return from the chamber was then gated. When I got the sound that I wanted, I put it into an AMS DMX 15-80, which then was triggered by a Linn drum machine. I treated the snare in a similar fashion.

"Setting the gate for a sound like this is a difficult thing. The point is committing to the close-down — really listening to the drum-machine sound, and making sure that the gate is closing musically. Once you put it on tape, you're committed: if the sound is great and huge but the gate is a little off, it can ruin the track. I prepare for that by sampling sounds and putting them away, in case I've missed it." *. . . continued —*

ROOM AND MICROPHONE LAYOUT TO ACHIEVE AMBIENT DRUM SOUNDS DURING BASIC TRACKING.
Studio: The Village Recorder Studio D, Los Angeles
Engineer: Daniel Lazerus
Note the use of an echo chamber for mikinq of live room ambience.



Lazerus says that the room diagrammed above was used during the recording of basics for the forthcoming Rosie Vela album *Zazu*, produced by Gary Katz. Two Neumann U87s and a single AKG C414 were mounted inside the live chamber, with the door open about two feet. The 87s were positioned two and five feet from the floor of the chamber near the door, while the C414 was located eight feet inside the room and eight feet off the floor. He also points out that because of low-end build-up, a little compression of the chamber mikes may be necessary, or the outputs from a few distant mikes mounted in the tracking room blended in with the chamber sound.

For guitar-amp miking, Lazerus uses a SM57 two inches from the speaker at a slight angle to the cone; an AKG C451 mounted two feet away at the same height; a C414 mounted three feet away, and a foot or so higher; plus a U87 mounted five feet away.

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cycles. So it can seem that you have a lot of bass, and then you realize that you may have more than you thought. But it's an instinctive thing, too. I engineer, really, from the heart — as silly as that sounds [laughter]. I'm *always* putting myself on the edge with each new project; I don't want to stop learning.

R-e p: Do you have a preference in mixing consoles?

DL: Very much. I love the older Neve consoles — pre-Eighties, early Seventies. They have incredibly musical EQ: in the earlier Neves, the engineers just seem to have chosen very musical positions — it's step, not sweep, EQ. When I work on other boards, and maybe have the opportunity to have access to an outboard rack with Neve EQ, I'll run drum sounds and things like that through it.

R-e p: I'm also interested in your thoughts on digital audio. Do you have a preference in digital multitrack and mastering machines?

DL: To be honest, I'm not all that familiar with the currently available digital recorders. I've spent a year working on the 3M [DMS] 32-track machine and, probably because it's the machine that I'm most familiar with, that's the one that I've worked on most. Soundworks [New York] has that machine, and I've done several albums there. I like how I can punch-in, roll back, record, overdub, and bounce tracks on the 3M.

I have done a lot of work in the digital domain, however — *The Nightfly* is entirely digital [3M multitrack; mix to Sony PCM-1610] — and I've found that there's something missing in recording purely digital. It's almost like you can have something *too* pure, *too* clean. You listen to *Sergeant Pepper*, or "My Generation" by The Who — those sessions were done on analog two-tracks, or four-tracks, or coupled four-tracks. And there's something wonderful about the "analog sound" of those records.

You see, there's a way that I like to record, if I can. What I like to do is use an older Neve console, record on an A-800 Studer, use [Scotch] 226 tape at +6 elevated level, hit the tape *very* hard, and cut my basic tracks that way. Any chance I can get, I try and steer toward that approach. There's a tape-compression effect that happens — particularly with drums — and, recorded properly, it's so clean and strong. Then, I immediately transfer those tracks to digital.

For me, the digital domain works better in the overdub and mix stage. I still prefer using analog to track the basics, but I like to do all the overdubs on digital — which is a great way to do all that bouncing, vocal combinations, and so

. . . continued on page 41 —

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One feature of this system enables you to by-pass the Channel VCAs, thereby

— continued from page 36 . . .

Daniel Lazerus

on. You end up with basic tracks which, when you solo them, you hear a kick drum recorded hot and strong in an analog sense, and transferred — without generation loss — onto digital. The sound is *unbelievably* clean: you could never tell that it wasn't just recorded on the digital machine, but it has the quality of the things I love about analog. I mean, when I mix, I'll even rent up to six, two-track slap machines — rather than using digital delays or digital slap — because I just *love* that sound: there's just something great about genuine analog tape slap.

R-e/p: So you're using each technology — analog and digital — for the best of what it can do for you.

DL: Very much so. See, in working with . . . [Pauses, then smiles wryly] Okay, Donald [Fagen], here I go. *The Nightfly* was one of the purest digital recordings that has ever been done. It was never in the analog world — the album was purely digital, and painstakingly so. And it sounds good: if you go to any HiFi store, that's what they usually demo with. It's almost a joke! Yet, with all that in mind, we've cut a few tracks for Donald's new album, and done it in the analog-tracking-to-digital mode. The reason Donald is recording that way — rather than purely digital — is because of things that I've done since *The Nightfly*; he heard the difference.

R-e/p: Donald Fagen's work is notoriously well-engineered and well-produced — very snappy, very clean. It seems to me that this level of quality has to be the result of an attitude that runs through the whole recording process: miking, pre-amps, the type of console used, choice of outboard equipment, and so on. How did you achieve that level of sonic purity on The Nightfly?

DL: With an artist like Donald, it really is a matter of — every single day — a meticulous concern for how things *sound*. If something wasn't sounding right, there could be several days spent on continually trying to achieve a particular sound. *The Nightfly* represents a very strong, perfect recording: if you were able to solo the tracks, you'd find that they're very clean, and very pure. And having an objective mixer just helped the whole process: *The Nightfly* was mixed by Elliot Scheiner, and the mixing process was unbelievably involved.

I recorded the bulk of the album and, throughout the entire recording process, it was all about subtleties, like removing an "s" from the end of a word. Or — literally, this is the truth — punching in on one syllable and getting out before the next one! Of course, the great thing about digital is that it allows that degree of precision in punches. So, for someone like Donald, who would want, "Uh, can you get in before the 'er' and out before the 'th'. . . " or whatever, digital is *perfect*.

Donald is just a purist beyond your and my imagining. Which works for him, certainly, but in some ways I'm sure it also works against him. It's really just Donald: that's how *he* hears it, and he cannot do it any other way. Certainly, I've worked on records that have cost much less and sounded very good. And I also like things that are raw and recorded very live. But this is the way he works, and he's paying the bill. And he's right: you cannot listen to his records and not hear great music, and brilliance, and groove.

I'm very proud of having worked on that album. I sang on it; I did a lot of experimental miking ideas; and I stretched quite a bit. It was difficult, but rewarding. I could probably talk about it more if it wasn't so good.

R-e/p: How did you come to be involved with sessions for The Nightfly? *Was it through your work with producer Gary Katz?*

DL: I was working at the Village Recorder, and had already worked at several other studios: I started at Wally Heider [Hollywood]. When I began at the Village I was a second and, within a year, I was put on a project with Gary Katz and Roger Nichols — it was the debut album of a group called Eye to Eye. Roger had a lot of prior commitments, and after about two weeks he said, "You should just use this guy. I can't do it." He pointed to me, and I took over the helm. I finished the album, and was asked to work on *The Nightfly*, which was kind of a coup: nobody else had penetrated that world before, and I wound up being the basic recordist. So, it came about from Roger Nichols' insight about me, his generosity, and his lack of insecurity about involving someone new.

. . . continued overleaf —

ANNOTATED DISCOGRAPHY — *concluded . . .*

Various Artists: *The Gospel at Colonus.*

"This is a gospel show, composed by a guy named Bob Tilson. It's certainly experimental theatre; a sort of *Oedipus* set to gospel. Donald Fagen gave me tickets to a performance of the show at the Brooklyn Academy of Music, and I was so struck by it that I sought out the composer and talked to him afterwards. It turned out that nobody had made plans to record it. I approached Donald about the idea, and we took it to Warner Brothers.

"It's not necessarily the kind of music that's 'hit-bound,' but it is amazing. I mean, the crowd at BAM went nuts! There's this group, the Five Blind Boys From Alabama, and their lead singer, Clarence Fountain, who's blind, plays Oedipus in the show. Oedipus, as you know, blinded himself. Clarence can turn on the stage like nobody's business, and he sings his ass off!

"The music involves a soloist, a band, and a huge choir. Miking the choir was an intriguing situation. When we came to doing the choir overdubs, which involved about 45 people, we were working in a studio in New York [Soundworks] that was extremely dead: low ceilings, and kind of crowded. But this music was about the church! So, we had the full choir in a semi-circle in this space, and I miked them in three sections from in front and from behind with overhead mikes: Neumann U87s behind, and AKG 414s in front. I set up an EQ that seemed to work, blended the mikes, and recorded them in stereo. In the mix, I added a concert hall effect from an Eventide SP2016 digital echo system.

Lazerus and Harrison: "5 Minutes/Bonzo Goes To Washington."

"This is a thing I did with Jerry Harrison of the Talking Heads, recorded in a basement in Milwaukee about a half a year ago. It's like a street kind of dance thing. I play drums and keyboards, Jerry plays keyboards, and Bootsie Collins plays bass. Ronald Reagan does the rest: it's based on his famous 'blooper' speech in which he announced that he was 'outlawing Russia forever', and that 'we begin bombing in five minutes.'

"Reagan's voice, taken directly from a recording that we got from the Mondale campaign, was sampled into an E-Mu Systems Emulator II. The keyboard was split so that different portions of the sample were in different registers of the keyboard, and Jerry played it. This recording was done extremely quickly, and put out before the election. It made it to number 38 on the dance charts here, and was very successful in Europe."

John Denver: "African Sunrise," from *Dreamland Express.*

"This is John Denver at his sincerest; as we all know him. I love this track: it's very much a 'John Denver' song. John has been involved with the World Hunger Project for about 12 years, and I can attest that it's a *very* real thing with him. He wrote this song in Africa —about a year before USA For Africa — when he was in Ethiopia and living with those people. I did cricket overdubs on this track, and Roger Nichols does bird calls.

"This whole album was done with one band: two guitarists, an acoustic piano player, a sax player, and a drummer in a small booth. I'd generally been working in large rooms, with areas to displace everything, and this was a small room. Nonetheless, I attempted to get a live drum sound by putting two Neumann U87s in the booth, facing the glass. This was the only track on the whole album in which I could use those 'ambient' mikes. In the beginning of the song, the toms, played with brushes, are from the stereo room mikes." □□□

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Daniel Lazerus

I was born and raised in Southern California — in the city of Orange, in Orange County — and all of a sudden I not only found myself doing a Steely Dan album as my first album, but also found myself in Manhattan. It was just a whole new world — a lonely proposition in a lot of ways, but it was rewarding, and I got a Grammy nomination for it.

R-e/p: Let's talk about tracking. Do you tend to go for a specific sound that you want to hear in the final mix, or do you try simply to record strong tracks that you may refine later with EQ, effects, and so on?

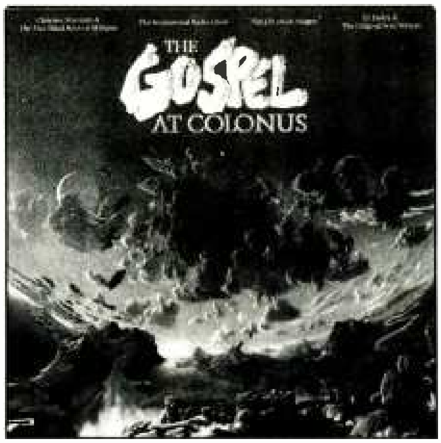
DL: When I'm cutting basic tracks, I try to make sure that what's on tape is *rock* solid, and has all the qualities that are necessary. And I drive myself crazy about that: I change instruments and amps around a lot, or rent other kinds of drums, because if the sound isn't there with the drums or the instrument amp, you *can't* fix it with radical EQ.

But I definitely go for a very specific sound. And it's difficult sometimes, because performance is *so* important as well. Nonetheless, I have a very strong, innate sense of what I'm going for, and I do have a style: there are some strong similarities that I drive towards from record to record, and from studio to studio. Yet, I want to understand the sense of performance.

So, in the tracking and overdub phases, I try to set myself up so that what I'm going to be mixing sounds great — so that it's right there at the end of the fader, and I'm not going to have to fool with it too much. The things that I worry about . . . [shakes his head] Sometimes, when I hear albums that I've finished, it's excruciatingly painful — if something doesn't seem too loud, then there's a shaker that wasn't loud enough in a certain section! I have to fight that, too, because these days cutting basic tracks is practically getting to be like mixing: you almost want to be able to cut your basic tracks and hear them like a final mix. That is quite difficult to achieve.

R-e/p: Especially since, at the outset, you don't really know what the texture of the final mix is going to be, since the whole sonic reality is unfolding before your ears as you work.

DL: Right. It's just the sophistication of the business that has caused it to be that way. But in the projects that I've done, if you put up the basic tracks, what you'll hear will sound like final-mix sounds. At times, I've almost degenerated my basic track sounds by re-EQ'ing things. They had been so well thought out to start with, that they didn't really need that!



R-e/p: Do you cut many effects during tracking?

DL: No, hopefully none — except for live drum tracking, and maybe a bass effect. I want to have the control that comes from having a good-sounding kick with no snare drum coming through it, a snare track where you can barely hear the hi-hat — that's all a matter of good miking technique — and then two faders with live, huge, stereo panned drums.

R-e/p: What miking configuration do you use for the ambient tracks?

DL: In the case of the last project I worked on, it was nine different mikes: room mikes, some mikes that were in a live echo chamber, and some of the direct mikes as well. A live drum sound that you can put on two tracks needs to sound *good*: it can't be off balance, or too "spikey," or have too much snare or cymbals. That takes technique.

R-e/p: Do you normally mix the basics that you track?

DL: Yes.

R-e/p: If you were not *to be involved in the mixing, would you still provide a stereo ambience track like that?*

DL: If it would be an added bonus for the person who was mixing, I suppose so. It's rare that I know beforehand that someone else is going to mix a project that I've tracked. And even if I was sure that was to be the case, I wouldn't be able to pre-guess their style. How could you know what they might want, unless you had spent years seconding for them? I just need to go with my instincts and, really — I have to say this again —a lot of what I do is instinctive.

R-e/p: Yet someone else mixed The Nightfly. *How did you deal with that situation?*

DL: I just tried to do my own quality recording. There's a lot of my personality in that album, just in the basics — things that I'm sure Elliot [Scheiner] wouldn't necessarily do. Recording the Hammond organ in stereo, for example, so that you can actually feel the Leslie speakers going around.

R-e/p: Can you describe that technique for us?

DL: The way I like to do it is to use two mikes in the vents on top. I use two different mikes, instead of the same kind, because it gives a different coloration and sonic quality. I use a Shure SM56 or 57, and an AKG C451 or 452 with a 10-dB pad. On the bottom, I use a [Neumann] U47 or an [Electro-Voice] RE-20 — or even a [Sennheiser] MD-421 would do. I pan the top mikes left and right, and blend the bottom mike into both channels. It's a *great* sound.

R-e/p: Let's turn to the mixdown process. Do you conceive of the final mix in visual terms?

DL: Very much so. I really see mixes like paintings framed between the two speakers. What I try to do is get the left-to-right panorama of that painting, but also create a three-dimensional feeling — as though you could almost walk into the mix, and grasp a specific sound. It's a matter of utilizing a lot of properly-controlled echo and delay: those are the elements that allow a lot of dimensionality, like a guitar that's here, with its echo over here [begins pointing in various directions]. A vocal that has its echo here, plus a delay that's sitting over here, and another delay that may be spiralling even further away, barely heard. Or a saxophone that's got an echo and a delay that are going this way; a shaker that's off here, but has an echo that's sending it off here. Things like that make the mix "breathe."

R-e/p: At what point in the remix do you start to bring in these various echo and delay elements?

DL: Well, I build all of the foreground elements first. The kinds of things that have spins and spirals are built on after the foreground parts, and they shouldn't be washed out or lost — bass guitar, in particular, is something that can readily be lost as the mix builds.

R-e/p: When do you first begin conceiving of this "painting" of a mix? Is it when you first hear a demo, during tracking, or when you really become involved with the mix phase?

DL: I concentrate on the basic tracks when I'm doing those, and take time to think about mixing before mix time. As far as the recording process goes, I do try to think about mixing in the sense of bouncing tracks properly; opening up tracks; keeping certain things open for mixing — things like that.

I don't always have the opportunity to hear a demo. When I do, an idea might spring to mind. But the concept of a mix really comes to me *after* the tracking sessions, when I spend time before the actual mixing begins, thinking about what I want to achieve. I make pages and pages of drawings of experiments that I want to try in order to make a

. . . continued overleaf —

Daniel Lazerus

drum go this way, or make a voice go that way; or make it bigger or smaller.

R-e/p: You said that, when you build a mix, you start with the foreground elements. Can you describe the process in more detail?

DL: To be honest, I really start by preparing all of the effects that I'm going to use. I generally use a snare drum to check my effects, because it's a regular, strong beat. I like to use three EMT plates — left, right, and center — and I make sure that they are balanced, decaying in the same time, and sound similar.

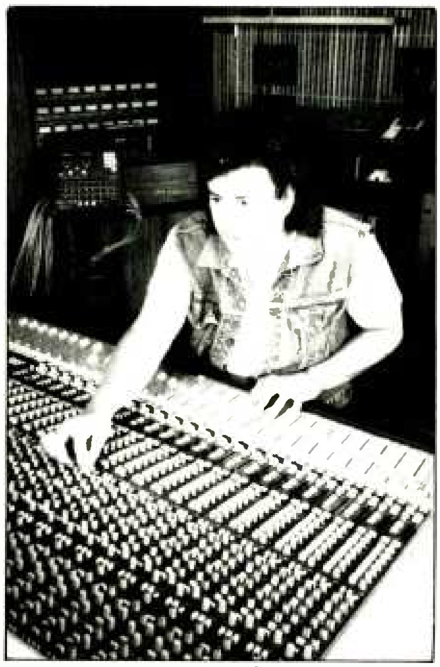
I check all the centers of my effects sends, then forget about all that. I start going for my drums dry — EQ'ing them in solo mode; same with the bass, guitar and keyboards. Then I'll put up a mix with drums and bass, and start to add the echo. Drums and bass are the foundation; I build from there, adding keyboards, guitar, and so on.

Even the drums might have certain effects, but I add those before I continue. And as far as setting delays and echo decays, to me that all depends on the drum sound. I'll solo whatever is the smallest increment of time on the track — usually, it's the hi-hat rhythm — and use that in setting delays. I'll listen to the return of the delay of the lead vocal, and make sure that it's just disappearing perfectly with that hi-hat. I love using tape-slap machines with VSOs so that you can hear a beautiful delay, but it just melts and disappears into the track. On a lot of my mixes, there are as many as seven or eight very different delays.

R-e/p: Have you had occasion to use sampling and replacement techniques?

DL: Oh, sure. While I was doing *The Nightfly*, I became familiar with Roger Nichols' computer, Wendell, which is the most sophisticated computerized sampling device that exists on the planet! Donald utilized Wendell to replace sounds on *The Nightfly*, and it was fascinating. Most of the time that we were working on that album, Roger was designing Wendell II, which was miles ahead as far as increased sampling rate, no loss of high-end, and so on. Then, Roger produced the John Denver album [*Dreamland Express*] that I just recorded and mixed, and we used Wendell a few times on that album, replacing the snare.

Now, on the album that I'm currently working on [June 1985] — which is an artist on A&M records named Rosie Vela — Gary Katz, who is producing, hired a guy named Jim Bralower. He's been doing Nile Rogers' drum machine programming, as well as for Hall and Oates and Cindy Lauper; he's just an



ace at programming drum machines. Jim studied all the demos, set up all his drum machine programs for each song, and we laid down the drum machine parts first. Then I worked with Jim Keltner — who played live drums on the album — on things like kick-drum sounds, using the live chamber at the Village Studio D. With Jimmy Bralower and an AMS [DMX-1580S], we were able to sample those live sounds and trigger them from the drum machine.

That sort of thing is almost becoming the norm. And I thought it was funny, because there was all this talk about, "Is Steve Gadd in Wendell? Is Jim Keltner's kick in Wendell?" Well, these days Jim Keltner will come to a date, and he's got all these weird drum sounds, noises, his Simmons toms, his Linn chips, the whole thing. He's beyond Wendell now! It's ironic.

R-e/p: You had a background as a musician before you began engineering, and on a lot of your projects you're credited as arranger, background singer, that sort of thing. How do you think your musical background comes into play during your engineering and producing?

DL: If I hear something that sounds wrong, I'll point it out. Or I might suggest a harmony part. On the last Joe Cocker album [*Civilized Man*] I arranged the percussion parts, and on Diana Ross' album [*Ross*] I arranged the vocal parts — by which I mean that I worked them out with the musicians.

I don't know that there are any rules about those things, but I don't want there to be any misunderstanding about terms: I *wasn't* the "music arranger." Any musician is usually free to develop his own parts, unless you're doing a jingle or something like that; it's rarely quite that strict. Even with someone like Donald Fagen, who knows very specifically what he wants, there's still room. That's important. I can't tell you all the great "mistakes" that have been on the records I've worked on. Just wonderful mistakes! Even on my part — by mispunching, or something like that.

R-e/p: I've always felt that the engineer — be it in the studio or in concert — formed an integral part of the musical ensemble. From what you say about your work, I suspect that you may feel the same way. You appear to have an intuitive approach to the recording process; you think of what you're doing in musical terms.

DL: Well, the musician's approach has been a useful tool for me in engineering, which I never really thought I would be doing. I was drawn into it. One example of how my musical background helps is that I can read charts. The artist or producer can say, "Go back to bar #34," and I can roll the machine back to that point. Certainly, some very technical people are good engineers, but I think it probably works better if they have a musical background.

R-e/p: But in crafting the sound of a track, you also are greatly influencing the emotional impact of the song.

DL: Well, I feel that about mixing, in particular. As a rule, when I mix, I'm left alone — and that's when I need to have the overview. And I feel that it is an incredibly important role: you really affect the musical result, unless you have a producer who says, "I want this, I want this, and I want this!" It's a difficult position, because you're coming up, and you also have working relationships with various producers to consider.

Yet a brilliant engineer can add *so* much to a track. I believe in the engineer/producer-to-artist relationship, and by that I mean in the sense of how Hugh Padgham or Bob Clearmountain are working. As a rule, an artist *knows* what they want, and sometimes production seems to be a matter of the extent to which you can get out of the way of their vision. I believe that a lot of artists desire a working relationship with someone who *bridges* the creative gap in the studio: a person that can capture the sound that the artist can only describe.

If I were to be producing exclusively, I would still want to handle the engineering side, because of the sense of control you have, plus the immediate communication with the artist and what they're working toward. I feel it's a valuable function, and I think that there are a lot of engineers who are very much involved in the records that we hear, yet they don't receive any credit or reciprocation. It's kind of like being a taxi driver: you're serving as the link between the passengers and their destination. I believe that the relationship between the artist and the engineer/producer, a co-production situation, is healthy for record-making. It's certainly what I desire for my career. ■■■

▲Clockwise from above: David Bowers, Dirk Schubert, and Ed Simeone

Photography by David Scheirman

TOTO World Tour

SOUND SYSTEM DESIGN BY SCHUBERT SYSTEMS GROUP

by David Scheirman

Take one of the music world's best-known, Grammy Award-winning pop rock groups, a custom-tailored concert sound system, 136 stage inputs, computer-controlled stage instrumentation, a new concept in monitoring, six (6!!) veteran live sound mixers, and several months' worth of international touring . . . and the recipe exists for one of the most complex concert sound projects to be out on the road this year.

Starting in February of this year, The Toto Entourage kicked off its 1985 World Tour in Japan. A complete stage-monitoring system, and a house-mixing package equipped with a regulated power distribution system, were shipped to all international dates. For the Japanese portion of the tour, a house reinforcement system was contracted through Hibino Sound. All North American dates were handled by Schubert Systems Group, of Gardena, CA.

On March 19, 1985, Toto's U.S. tour began at the Arizona State University Activity Center in Tempe, Arizona. This writer journeyed to the site for a first-hand look at the group's advanced audio system, which features multiple mixing consoles with six operators.

The Band

Toto is a notable group, comprised of some of America's busiest working studio session musicians. The group's albums have often featured innovative recording techniques and instrumentation. Collectively, members of Toto — David Paich, Steve Lukather, plus Jeff, Steve and Mike Porcaro — have probably participated in the playing, arranging and recording of more recent, American popular rock music than any other similar group of musician/technicians.

Several of the band members operate personal-use recording studios, which are stripped of gear when the group does one of its infrequent tours. Shep Lonsdale, a recording engineer and audio mixer who has collaborated with the goup on such recent ventures as the film soundtrack for *Dune*, is involved in all aspects of Toto's sound.

"Having been involved with the actual recording to Toto's music in the studio gives me a much different perspective on doing the live shows than many concert mixers might have," explains Lonsdale (*pictured left*). "Traditionally, a gap has existed between live and recorded sound.

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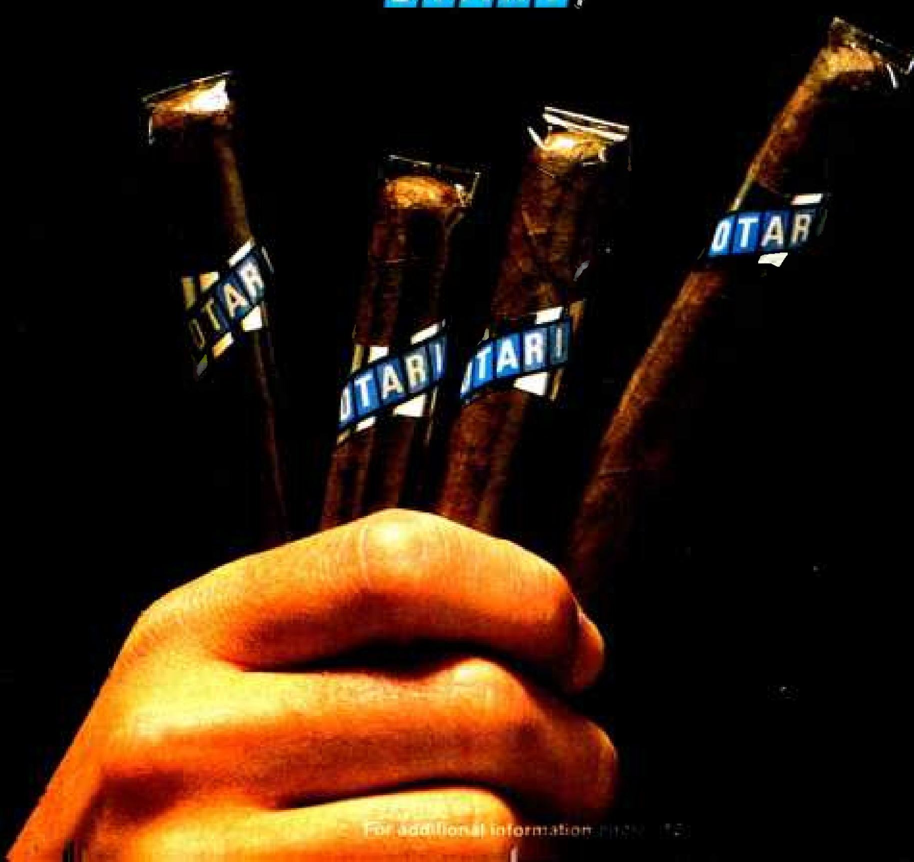

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TOTO WORLD TOUR

That gap is starting to narrow, as the technology becomes available to recreate the sound of an album in a live-performance setting. Stage technology is improving, and concert-sound systems are now beginning to offer the fidelity that has been lacking in the past. The sound of the recorded music *and* the sound of the live show are important to the members of Toto."

The Live Concept

Dirk Schubert, of Schubert Systems Group, was actively involved in assembling Toto' s custom studio monitoring system and performance hardware. When the band chose SSG to provide full sound reinforcement services as well for the group's live tour, Schubert went along as a monitor mix engineer.

"One of the most important things to understand here is that these guys *know* what they want," states Schubert (*pictured above right*). "They craft their musical packages in the studio, and they are used to a certain way of hearing everything . . . stereo keyboard monitors, special vocal monitors, and instrument submixers. It was up to *us* to figure out how to take the whole thing on the road in an easily-transportable package.

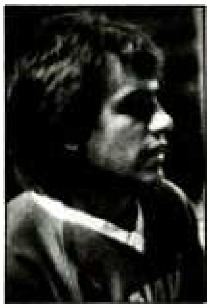
"Our touring accounts get something that other sound companies are not often able to address: if the gear doesn't exist to give them the sound or the operational functions that they want, *we* build it for them."

To make Toto's live show happen, 136 stage inputs were funneled down to house and stage monitor mixing positions, each with two Gamble consoles (and two board operators) by using separate, manned stereo keyboard submix positions (Figure 1).

Hidden offstage, these two consoles gave both keyboardists David Paich and Steve Porcaro an individual audio mixer for submixing the multiple stage rigs (with banks of MIDI-connected keyboards), as well as a "private" stage monitor man for each musician's own musical program material. Personal computers assisted in the MIDI-switching of the two rigs.

Keyboards; Stage Left

The instrument submix for Steve Porcaro was handled by Ed Simeone, who gained a great deal of experience with complex keyboard setups during his several years of touring with Electric Light Orchestra. Simeone mixed on a Gamble SC24-24-II board. Designed and built by Jim Gamble Associates, the transformerless console features 24 inputs and 24 outputs, and seemed to be ideally suited for use as a stage instrument mixer (Figure 2).



"Basically, what we are doing here is giving the musician access to a wide variety of electronic instrument voicings, and using MIDI-switching technology to keep stage clutter at a minimum," Simeone explains (*pictured left*).

Several primary keyboards are located on stage at Steve Porcaro's position, including a Yamaha DX-7. Interface cabling connects the stage area with Simeone's setup, and those performance keyboards can access the additional instruments, including a pair of E-mu Systems Emulator IIs and an Oberheim X-Pander.

"Some of the latest electronic keyboard gear now available is being

Figure 1: Stage Diagram of Toto 1985 World Tour.

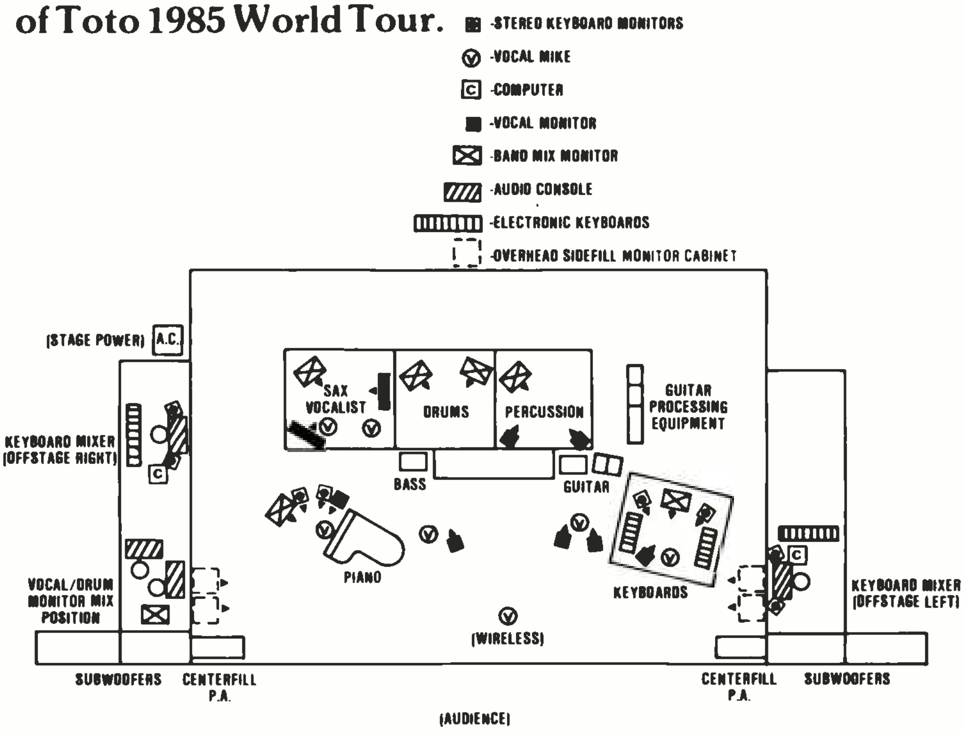


Figure 2: Stage-left keyboard mix position — Gamble SC-24-24-II console, manned by Ed Simeone.

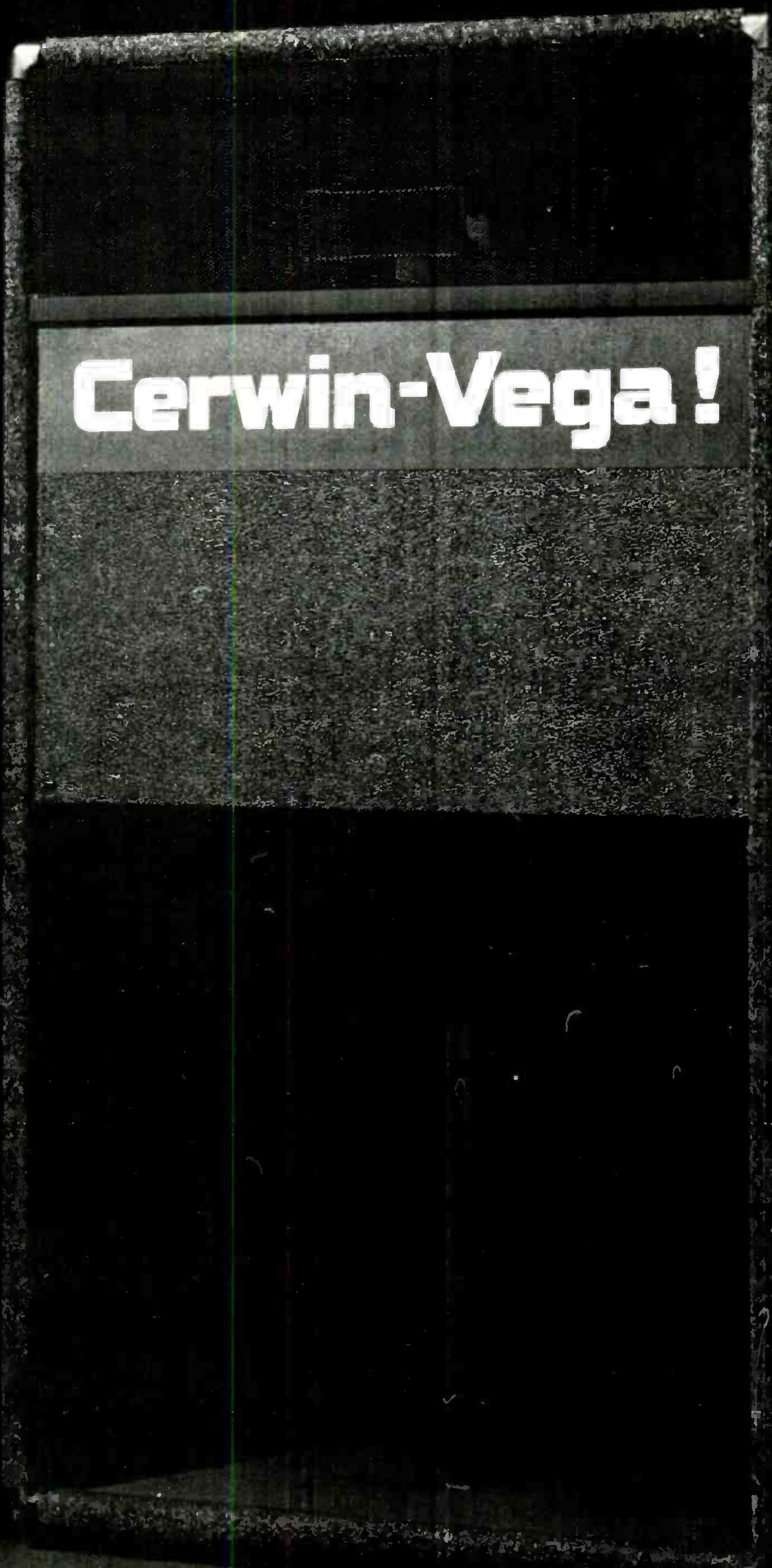


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Figure 3 (*left*): A Compaq personal computer located at the stage left submix position. Each song in the show has a "menu" showing which instruments are in use, as well as MIDI patches. Figure 4 (*right*): The Compaq display screen, showing an informational display designed by keyboard expert Ralph Dyck.

used with this show, and the factory support from companies such as Yamaha, Oberheim and Emulator has been tremendous," Simeone confides. "Things could get pretty complicated with this many keyboard units." Having the auxiliary keyboard racks off-stage gave the stage a much cleaner appearance, he offers.

To help the complex setup work smoothly, Simeone uses a Compaq personal computer placed next to his mixing desk (Figure 3). The screen editor displays a "menu" for each musical arrangement, showing what keyboard device is patched through which MIDI switch for each tune (Figure 4). In addition, the computer program (arranged by keyboard expert Ralph Dyck) sends a pulse to a JL Cooper MS-II MIDI match-box device, when authorized by the engineer. Yamaha MIDI rack panel modules are hooked up to the various keyboard devices. A customized voltage-controlled attenuator module was designed by Jonathon Little of Village Recorder, Los Angeles, to provide a direct interface for lever-changing of the keyboard instruments.

"This is a *very* high-quality way to control levels," noted Simeone. "It would be counterproductive to put a very clean signal from a $50,000 console through a $1.50 volume pot! Steve Porcaro uses the VCA to control his whole rig, while David Paich uses his to fade different synthesizers in and out of the piano mix."

Effects devices available to Simeone included a Yamaha REV-1 digital reverberator, a DL-1500 digital delay unit, a Roland Super Jupiter, and a Lexicon Prime Time II digital delay. In addition, a Dynacord CLS-222 was available for an electronically-created Leslie rotor effect.

The entire keyboard mixing rig was streamlined, and seemingly well-designed as a synergistic package. All inputs and outputs to and from the VCAs are patched, and a 42-pair multicable connects the synth rack and console. For fail-safe operation, a 16-channel manual switching panel can take over in case of MIDI "hang-up." Additionally, a MIDI "Panic Button" is supplied just in case a glitch in the complex control-signal path line causes the system to disregard a computer instruction to change over to the next song's settings.

"On this side of the stage, Steve likes to wait until I do the changes, and then he kicks it over himself with an on-stage switch," explains Simeone. "On stage right, I think they did have a hangup once or twice during the shows in Japan, but a quick tap of the panic button sent out a burst of signal pulses in about 30 milliseconds, and that cleared it up. The button interrupts the signal bus, and gives the circuits a chance to clear."

Keyboards; Stage Right

David Bowers, who has worked with the Doobie Brothers and Kenny Loggins, among others, mixed David Paich's stage-right keyboard rig. Bowers also commanded a Gamble SC24-24-II desk that was located off-stage right (Figure 5). Here, a bank of MIDI-connected keyboards was directly controlled with an IBM personal computer (Figures 6 and 7).

An Oberheim DSX synthesizer with Digital Polyphonic Sequencer, Ober-

Figure 5 : Stage-right keyboard mix position — Gamble SC-24-24-II console, stereo monitor loudspeakers, and an IBM personal computer. (This position manned by David Bowers.)

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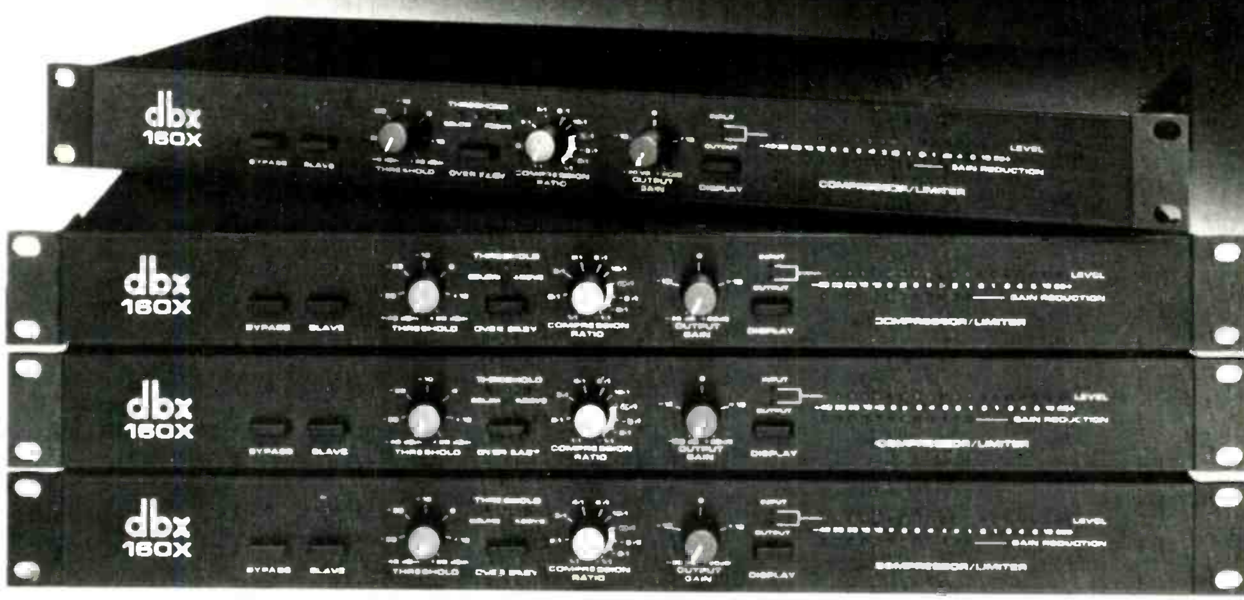

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Figure 6: Stage-right auxiliary keyboard rack, where MIDI-connected instruments for David Paich include an Oberheim DSX, Oberheim X-Pander, and an E-mu Systems Emulator II.

Figure 7: David Bowers' IBM computer handles MIDI switching for each tune during the show.

Figure 8: A Yamaha DX-7 synthesizer has been neatly fit into the top of a Yamaha concert grand piano. The DX-7 and piano each access the off-stage keyboard rack.

heim OB-8, Oberheim X-Pander and Emulator II were all controlled by either a Yamaha DX-7 or Paich's concert grand piano. A cut was made into the piano frame, and the DX-7 nested snugly on top (Figure 8).

"The computer, the keyboard audio mixer, and MIDI technology make things a lot easier on stage than they used to be," states Bowers (*pictured right*). "Instead of mountains of keyboard instruments and miles of spaghetti-like cables, you just see a man up there with his piano. But, you are hearing many of the exact voicings and synth parts that appeared on such classic tunes as 'Africa' and 'Hold The Line'. The grand piano keyboard can trigger sounds that have been stored in the Emulator, which were taken directly from the album masters."

Bowers used a Lexicon PCM60, Roland MKS-80 Super Jupiter, two Yamaha D1500 digital delays, two Roland SRE-555 Chorus Echos, and a Lexicon 224 digital reverb for special-effects processing. In addition, an Aphex Compellor compressor-limiter and an Eventide H949 Harmonizer were available in the equipment racks.

As on stage-left, a JL Cooper MIDI Match-Box and a custom VCA panel formed part of the setup, along with a Yamaha MIDI Rack.

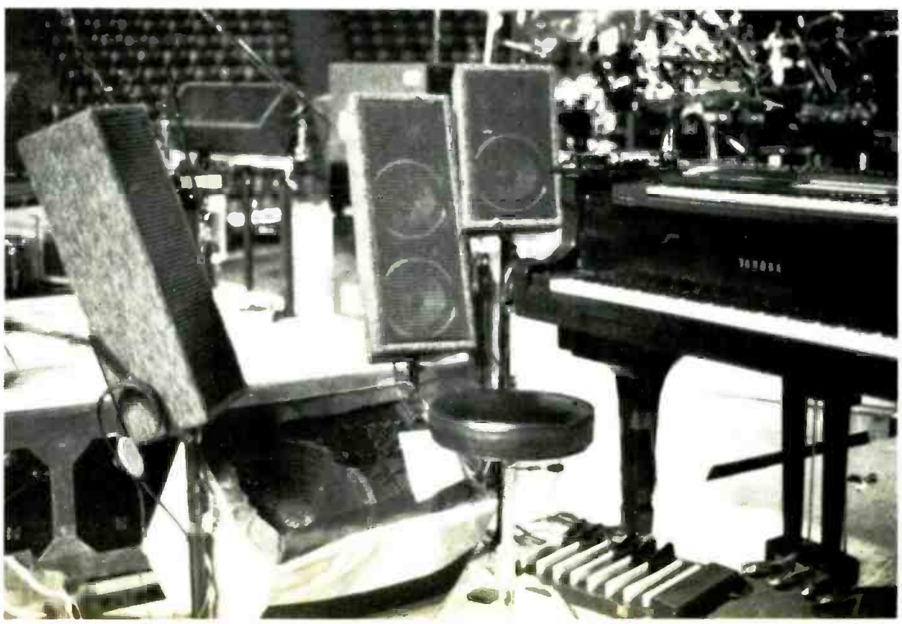
Keyboard Monitors

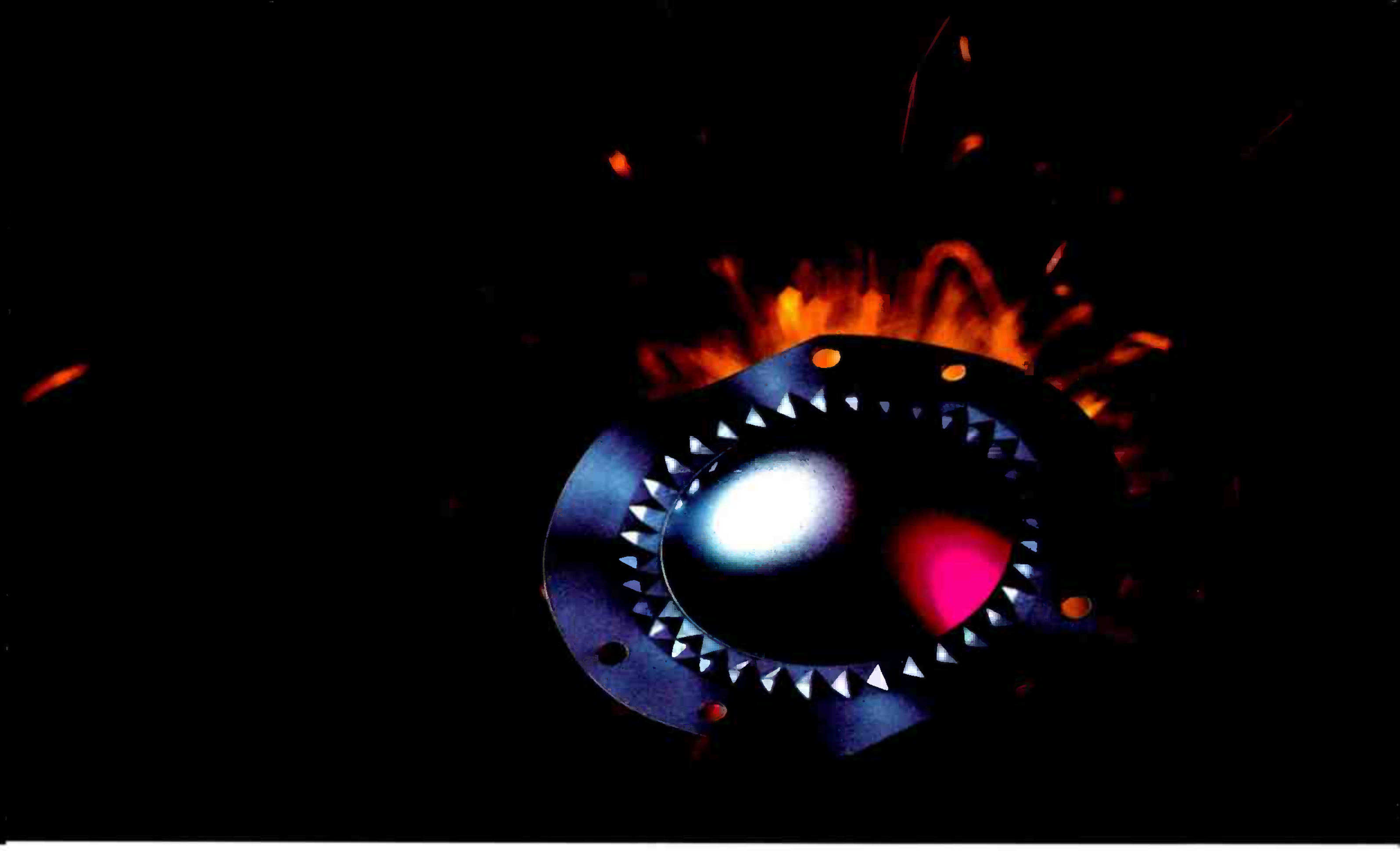
"An important part of assembling this stage-monitoring system was the concept that the performers wanted small, bright-sounding boxes placed up at ear level," explains designer Dirk Schubert. "Also, nobody in this band wants to hear much of anything below 150 Hz on the vocals coming from these boxes. It is like a 'close-field' mini-monitor approach. What we basically had to come up with was the Yamaha NS-10 or JBL 4401 speaker concept that could put out concert sound pressure levels, *and* be able to hold up on the road."

Sets of compact, custom-built stereo keyboard monitors and interface electronics were designed and assembled by Schubert Systems Group to present the complex mixes to the performers. For console monitoring, both David Bowers (stage-right) and Ed Simeone (stage-left) used a pair of cabinets that were identical to those placed on stage.

The miniature loudspeaker columns each house two JBL Model 2118H eight-inch speakers with a passive contour network on each, and a 2404-H tweeter. The eight-inch speaker's frequency response is essentially flat from 150 Hz to 4 kHz, at which point the tweeter is brought in with a passive crossover network. The boxes are trapezoidal in shape, and fitted into the stage set with small metal support stands (Figure 10). Yamaha PC-2002

Figure 9: David Paich's stage monitors: the small enclosure next to the piano keyboard carries a vocal mix only. A stereo keyboard mix is also available from the mini-columns.





JBL's unique titanium diaphragm and "Diamond Surround" bring new purity and consistency to high frequency response.

Figure 10: Small loudspeaker columns, each housing two JBL Model 2118-H eight-inch monitors for Steve Porcaro on stage-left.

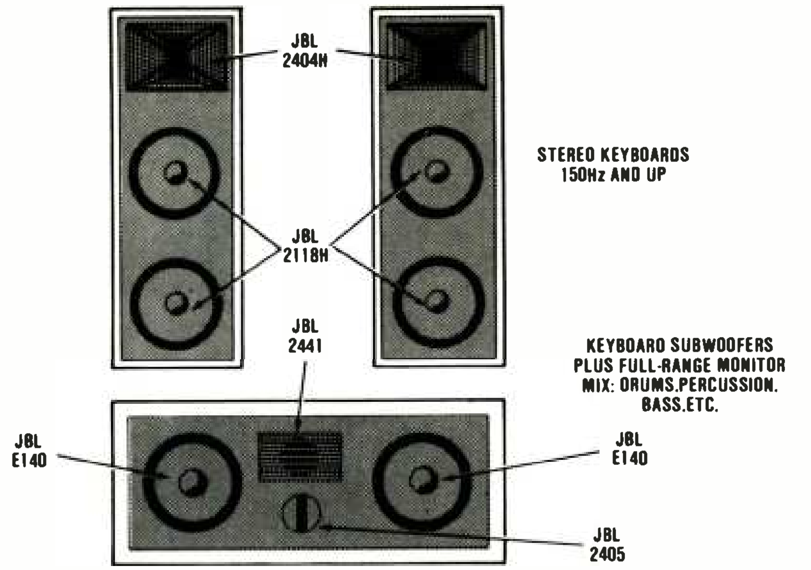
TOTO WORLD TOUR

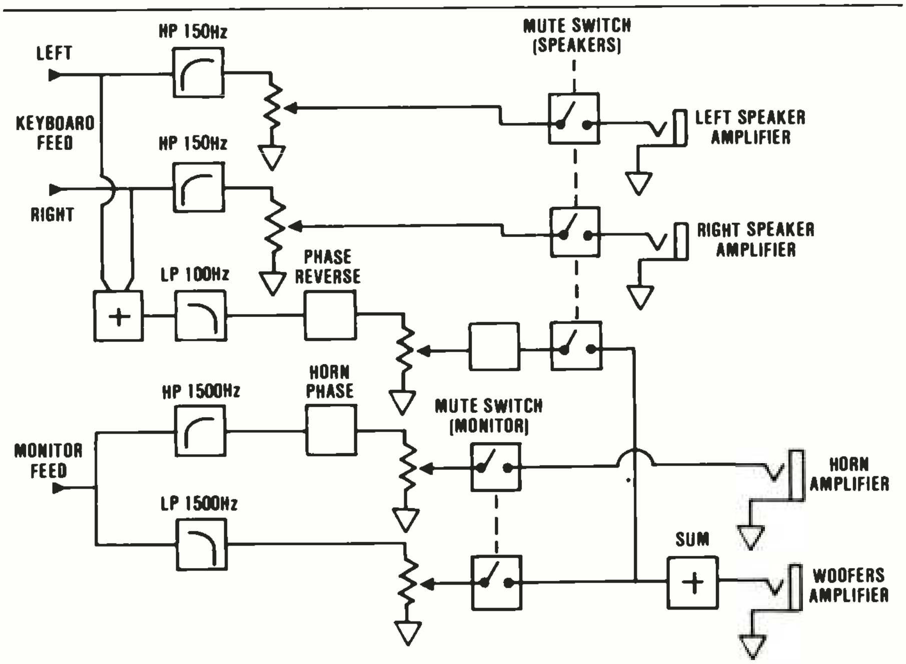
amplifiers power the keyboard rigs, while Metron A-400 amps drive the vocal monitors.

The small keyboard columns proved so popular during rehearsals that other performance areas also were supplied with them, including the sax/background vocal riser. "The concept really makes sense," explained sideman Scott Page. "The little boxes give us bright reference information to sing with, right there in front of us. The kick and bass sound, the main rhythm section mix, comes from a little farther away in the bigger slant, instead of blasting me in the face like a lot of stage speaker systems do. It works *great*."

Critical keyboard, vocal and solo-instrument program information is fed through the compact speakers. Additional rhythm section material requiring better low-frequency presentation is fed to the various performers through separate larger floor-slant monitors that house JBL K-140 15-inch speakers, 2441 drivers with 2445 diaphragms, and 2405 tweeters.

Where floor monitors are required for vocals, including Steve Lukather and Mike Porcaro, SSG's low-profile vocal slant monitors were used (Figure 11). These tiny boxes pack a pair of JBL 2118-J eight-inch speakers and a 2425 one-inch compression driver mounted on a modified JBL 2344 Bi-Radial horn. Actively crossed over at 1.5 kHz, the cabinets sit hardly 12 inches high, and offer an extremely smooth, yet bright, vocal reference mix; they also have a power contour network on the horn. A protective





Top: Schubert Systems Group Stereo keyboard Monitor Rig.
Above: Block Diagram of Keyboard Monitor Signal Flow.

cover latches into place for travel, and wheels make moving the package very easy (Figure 12).

Monitor Mixers

The main monitor mixing area (down-stage right) was handled by Dirk Schubert and Alan Bonomo (Figure 13). A Gamble SC40-16 served as the primary board, while an SC32-16 was used as a drum and percussion submixer. (The latter also served the opening act). What started as 136 discrete stage inputs ended up as 58 combined channels at the house and stage monitor positions, plus various effects returns.

Yamaha Q-1027 third-octave graphic equalizers were available for some of the 16 monitor mixes, although the Gamble boards feature on-board parametric equalizers across each output mix. A Lexicon 224X reverb with LARC remote, Yamaha REV-1 digital reverb, Lexicon Prime Time, Eventide H949 Harmonizer, and a Roland SDE-2000 digital delay unit were available for processing use on vocals, drums, keyboards and saxophone. dbx Model 160 and 160X compressor-limiters were channel-inserted for lead vocals, background vocal mix, piano and kick drum.

Flying overhead stereo, tri-amped sidefill cabinets flanked both sides of the downstage area. Lead singer "Fergie" Fredricson, using a Nady 701 wireless microphone, does not rely on any floor slants. The cluttered look of a half dozen slants along the front of the stage is changed here to a completely wide-open performance area.

"We have been using one of our PA cabinets as a side-fill box on each side, hanging from the lighting truss," notes Schubert. "The stage-sound level on this tour is much lower than it

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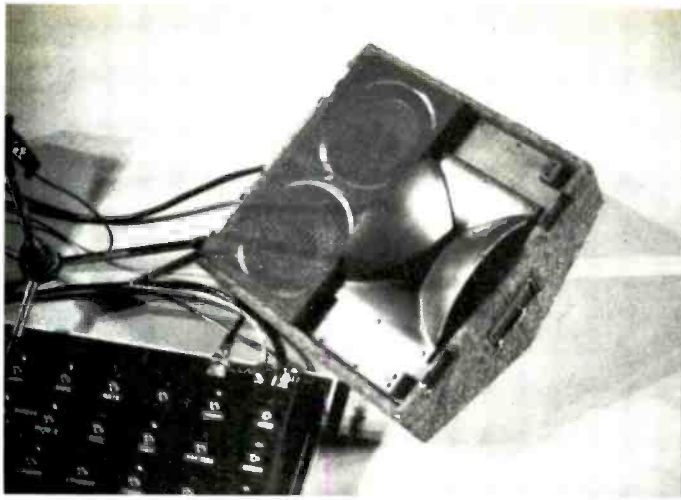


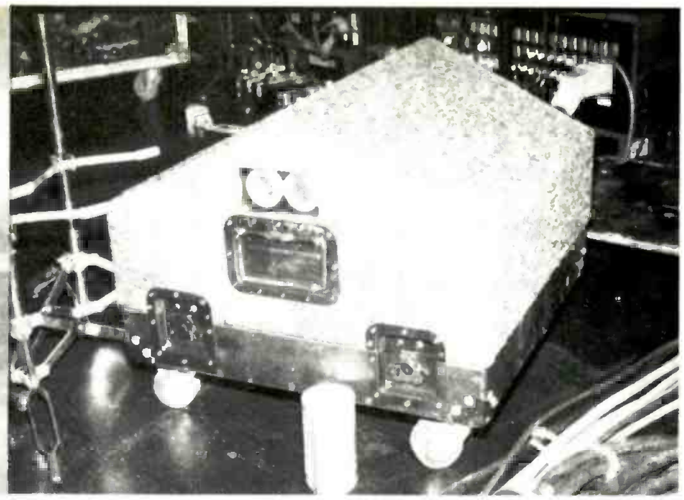


Figure 11 (*left*): SSG's custom low-profile vocal monitors each house two JBL 2118-J eight-inch speakers, and a 2425 one-inch compression driver on a modified Bi-Radial horn. **Figure 12** (*right*): A protective cover latches onto the mini-monitors for travel protection.

has *ever* been: it is about 6 to 10 dB down from when we used a traditional loud monitor system. With less sound up here on stage, we are finding that *everyone* hears more clearly."

Of Schubert's 16 monitor mixes, two went to the tiny floor slants; three mixes fed the miniature keyboard speakers; and five went to full-sized 15-inch slant monitors as rhythm mixes. Additionally, three mixes were used as effects sends for the vocals and drums, while a headphone mix was sent to the piano area, and stereo sidefills completed the monitor board's output assignments.

"Toto has been using the Gamble boards in the recording studio," notes Alan Bonomo (*pictured left*). "Since this whole complex setup has been created around the keyboard submixers and the different types of monitor cabinets, we are duplicating that on the road so the performers have the same system that has worked well for recording."

"It is important to note that a monitor system designed around the needs

. . . continued overleaf —

Figure 13: The monitor mixing area, downstage right: Gamble SC40-16 and Gamble SC32-16, handled by Dirk Schubert and Alan Bonomo.

TOTO WORLD TOUR

of a recording studio seems to be working well in a live performance situation," Schubert explains. "The rolled-off low-end, the smaller cabinets, the lack of floor slants for the front singer . . . it has all helped to cut down the stage noise tremendously. Things sound *very* clean up here."

Vocal microphones comprised Shure SM78, Beyer M88 and a Nady 701 (fitted with an SM87 capsule). The drum kit featured a host of Sennheiser MD-421s, while Countryman Isomax II miniature condenser mikes picked up the congas, bongos and timbales. A hybrid Fender/Yamaha wireless body pack unit was installed on the saxophone to allow freedom of movement.

International Tour Package

The North American concert dates posed no particular logistical problems for Schubert Systems Group, since the firm regularly handles nationwide touring assignments for a variety of clients, including the Tubes, Willie Nelson, Michael McDonald and Jefferson Starship. However, much thought was given to the many concert dates to be performed in Europe and Asia.

"Toto wanted the entire stage instrument package and monitor system to be self-contained and consistent," recalls SSG's David Morgan. "Due to the great number of signal processors, crossovers and amplifiers, it was important that all of the racks be standardized, while travelling as compactly as possible."

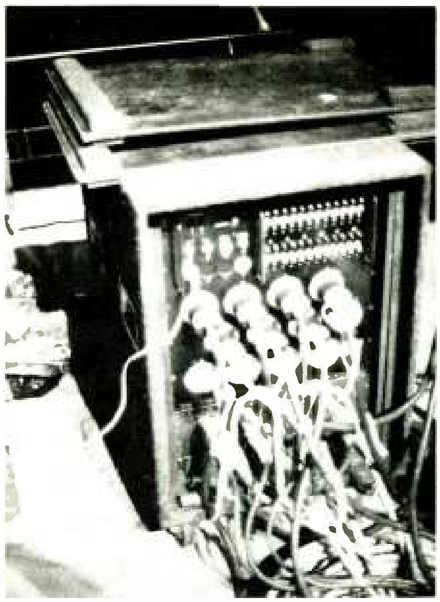


Figure 14: A regulated power distribution system offers each mix and stage equipment area two 20-amp legs of AC power.

The standard-sized electronics racks were fabricated by Flag Systems of thick birch plywood, and covered with a tough charcoal-gray exterior nylon carpet material. An inner, foam-surrounded birch frame protects the delicate electronic equipment. The racks measure 30 by 24 inches, and fit either three across in a 90-inch truck, or four across in the new 99-inch trucks.

Due to the microprocessor-based functions of many electronics devices, a clean, consistent source of AC power was considered essential. A compact regulated power supply was designed and fabricated by SSG (Figure 14).

"This distro serves the stage area, the monitor system and the house mix area," explains Schubert. "Each performer and console area has two 20-amp legs of clean, regulated electrical power. Every man is on his own breakers. If the AC starts to drop or surge, the regulators automatically compensate, and can be set to allow

Figure 15: House mix engineer Clive Franks at the Gamble HC40-24 console. A Yamaha M1516-A submixer handled drum and percussion inputs.

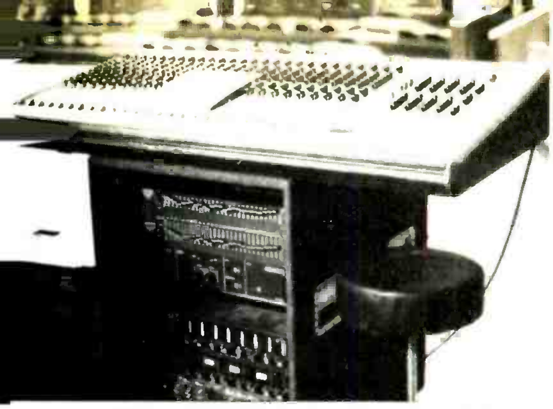


Figure 16: A separate Soundcraft Series 400 desk was provided for use by opening acts.

up to a 12 percent 'window' for the optimum voltage lever."

A custom-designed stage input panel/splitter system was assembled for the group, with separate record/broadcast capabilities for taking 96 lines on-stage into two 48-pair snakes.

A variety of unexpected difficulties can arise when taking such a complex live show to other countries. "We got to Japan, and were not even able to use our new Nady 701 wireless system because it turned out to be right in the middle of a Japanese television station frequency," Schubert recalls "Over there, however, products are available for use which cannot be purchased here in the States."

House Mixing

Shep Lonsdale and Clive Franks share mixing duties for Toto. The primary mixing console was a Gamble HC40-24, and a Yamaha M1516-A submixer was set up to receive drum and percussion inputs (Figure 15). A separate Soundcraft Series 400 desk was provided for use by the opening act (Figure 16).

Effects processing devices included a Lexicon Prime Time II, AMS 15-80S and RMX-16 delay units, Yamaha REV-1, Lexicon 224X digital reverb, and an Eventide H949 Harmonizer. Ten Valley People Kepex II noise gates were channel-inserted for drum and percussion inputs. Channel-inserted compressor-limiting for vocal microphones was assigned to dbx model 160 and 165 devices. Four Yamaha C200 stereo cassete decks also were supplied for taping the sh w (Figure 17).

"This is my first time using this particular sound system," explains mixer Clive Franks, known for many years of touring with Elton John. "It's pretty exciting. One can get better live sound results from a custom-tailored and correctly-engineered system such as this one. It's good to have the designer out here with us, though . . . [Dirk Schubert] — that makes things go more smoothly, since some of the devices such as the crossovers are not off-the-shelf, familiar products."

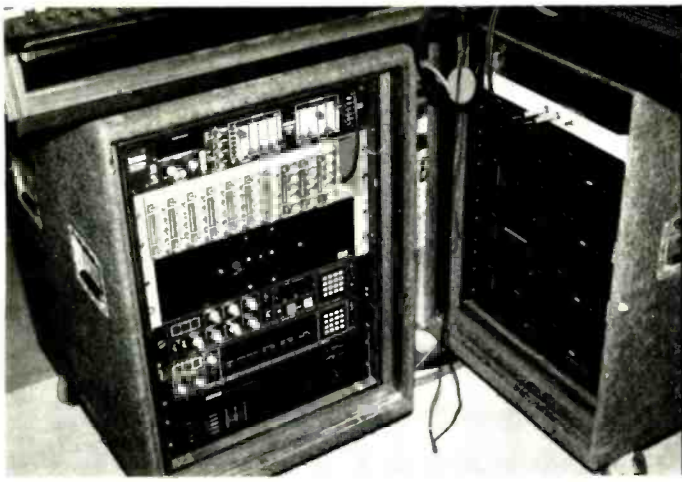
Like Shep Lonsdale, Franks felt that sound systems for live-concert use have been improving over the years. "We seem to be getting more sound from a fewer number of cabinets than what you would have seen several years ago," he notes. "Improved array design and increased amplifier performance are all part of it."

Lonsdale concurs: "Years ago, we made the best out of whatever we had. If you were good at what you did, you learned how to get the best sound out of anything, because so many of the available systems were poor in quality. It's pretty easy to find good systems these days, as we all keep learning about what it takes to do the job right."

House System

Schubert Systems Group's loudspeaker arrays comprise multiples of a three-way rectangular "column" cabinet, each of which houses two JBL Model 2220 15-inch speakers, a Bi-Radial horn with a two-inch compression driver, and four JBL 2402 tweeters. The cabinets are easily assembled into hanging arrays (Figure 18).

Large subwoofer cabinets, each housing four JBL Model 2245 18-inch loudspeakers in a ported rectangular



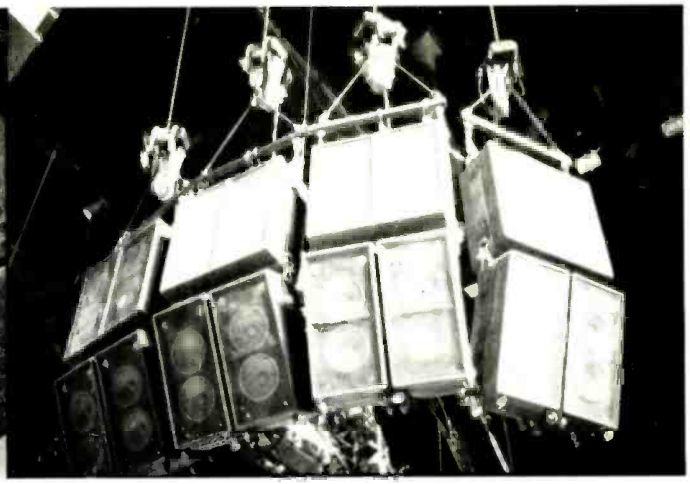


Figure 17 (*left*): House equipment racks held a variety of signal processing devices, including a Lexicon Prime Time II, AMS 15-80s and RMX-16 delay units, and dbx compressor-limiters. Four Yamaha C200 stereo cassette decks were available for making reference recordings of each concert. Figure 18 (*right*): A total of 36 three-way loudspeaker enclosures, each housing two JBL 2220 loudspeakers, a Bi-Radial horn with 2441 driver, and four 2402 compression tweeters, were supplied to the tour.

box, provide low-frequency reinforcement below 100 Hz. Stacked on the floor next to the stage, ramps also allowed these boxes to serve as an additional performance area for the acrobatic lead singer (Figure 19).

Amplifier racks house five stereo units each. Three 1,200-watt, one 800-watt and one 400-watt specially-modified Cerwin-Vega amps are currently employed, a combination that yields 300-watts to each 18- and 15-inch speaker, 150 watts to each 2441 driver, and 25 watts to each 2402 tweeter (Figure 20).

"Having enough amplifier headroom to adequately drive the loudspeaker system and the reserve to respond to transient peaks is very important to us," observes SSG technician Mike Ferrara.

A 200-amp, three-phase power distribution system drove the C/M Lodestar hoists used to "hang" the sound system, and supplied the main amplifier racks. A neat, modular I-beam system with heavy nylon straps suspended the speaker arrays; one rigging point with a one-ton motor suspended a single beam and four speaker cabinets. For venues averaging 10,000 seats, SSG supplied Toto with 36 three-way cabinets and eight subwoofers, giving a total of 32 18-inch

Figure 19: Large subwoofer cabinets, each containing four JBL 2245 18-inch speakers, provide low-frequency reinforcement below 100 Hz. Stacked next to the stage, the top surface of the cabinets offered an additional performance area for the acrobatic lead singer.

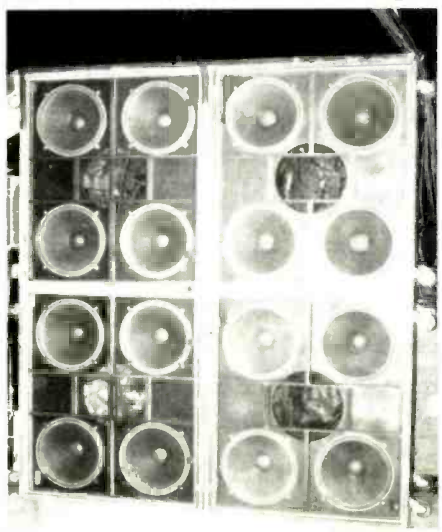


Figure 20: SSG amplifier racks housed three 1,200-watt, one 800-watt one 400-watt specially-modified Cerwin-Vega power amps; 300 watts of power is available to each 18- and 15-inch speaker.

speakers, 72 15-inch speakers, 36 two-inch drivers and 144 compression tweeters.

Conclusions: Performance Sound

With enough consoles and digital signal processing gear in this one touring system to fill a couple of audio rental supply houses, one begins to wonder where the trend towards extensive hardware for live-performance use will stop. A concert sound setup such as this one is extremely costly, and is far beyond the average system on the road today in terms of its complexity.

However, the extra care taken to assemble the audio tools required to achieve live duplication of recorded music deserves more than a few compliments. The concert that this writer attended at the Arizona State University Activity Center featured an extremely well-crafted mix, with subtle nuances and effects not often heard in live rock concert settings, particularly of the one-nighter variety.

The stage-area submixers were perhaps instrumental in achieving the excellent end result. The SSG system presented the detailed mix to a lively college-age crowd with power to spare. Full-frequency coverage was well-distributed throughout the listening area. Twenty years ago, a "rock and roll" show had one soundman, perhaps 12 stage microphone inputs, and whatever house-sound cluster was available that night. As I sat and listened to six experienced board operators mixing down 136 inputs on $300,000 worth of consoles and effects into high-fidelity hanging speaker arrays, the distance that the concert sound industry has traveled in those two decades was remarkable to behold. Now, if we can only "fix" those sporting-arena acoustics! ■■■

VISUAL MUSIC SCENE

The Growing Impact of the New Medium on Today's Recording and Production Studio Industry

Adrian Zarin reviews recent Audio-for-Video Developments

John Waite's "Restless Heart"

Elizabeth Annas/Photosensations

David Bowie's "Blue Jean"

Cats' "Rum Tum Tugger"

Looking back over recent Music Video developments, it's hard to avoid the conclusion that in several areas this infant industry has grown tremendously. On the artistic front, we've witnessed Music Video's evolution as a narrative genre through the incorporation of dialog, sound effects, and an overall directorial approach that is much indebted to feature films. Aesthetically, it's clear that video clips still have a long way to go. In particular, directors need to solve the problems of integrating two distinct art forms: music and storytelling. But, by incorporating basic narrative structures into their clips, directors have brought Visual Music into a new era of sophistication. For confirmation of this new era's arrival, one need only consider the impact of such productions as David Bowie's 20-minute video *Jazzin' for Blue Jean*; as an especially effective blend of music and narrative, *Blue Jean* stands as a promise of good things to come.

In many respects, however, Music Video's dramatic growth on the business front has all but overshadowed the artistic strides made during the last year or so. In broadcast, the past year brought many new Music Video outlets to both cable and the networks. We even saw several contenders rise up and challenge MTV's autonomy as our *only* 24-hour source of Music Video. While the most formidable of the challengers — Turner Broadcasting's projected 24-hour cable Music Video network — never made it past the first round, it still seems that MTV's title as video's sole tastemaker has passed into history.

MUSIC TELEVISION™

And while all this was taking place, Visual Music was becoming more and more of a big seller on the home videocassette market, via concert tapes, clips collections from leading artists, and rock-related feature-film releases.

So while directors and their creative teams pushed Music Video forward as a valid entertainment form (with value outside the promotional realm), business forces in the industry were hard at work creating more and better ways to bring Music Video to the public. Interestingly, both of these developments posed new challenges for the audio side of Music Video production, that oft-neglected dimension that has been the special concern of *R-e/p*'s regular "Visual Music Scene" columns.

The narrative use of dialog and effects posed new problems for audio engineers in terms of synchronizing sound and picture, while incorporating these extra aural elements into the soundtrack without encroaching on the music. And in both the broadcast and home markets, audio quality became a hot issue. Complaints were heard from programmers and viewers alike regarding the poor audio quality of the video clips broadcast on network TV and, occasionally, those sold for home use. The advent of Stereo Television has made the need for high-quality broadcast audio all the more dramatic.

Meanwhile, one of the newer 24-hour music channels, the Discovery Network, which planned to begin broadcasting June 1, will utilize audio as a leading selling point. Its organizers are eagerly publicizing plans to transmit the three-dimensional effect of Holophonic sound on UHF. (See the December 1984 "Visual Music Scene.") MTV, for its part, began distributing digital stereo audio to affiliates on January 1, via Dolby's new Adaptive Delta Modulation system. And on the home front, LaserDisc players, Beta Hi-Fi, and the new VHS Hi-Fi format have created a demand for home cassettes with audio quality comparable to, if not better than, records and audio tapes.

As a result of all these developments, audio professionals working in the Music Video field have seen a gradual, but undeniable, increase in audio awareness and sophistication on the part of their colleagues. The audio concerns of the contemporary Visual Music scene have affected everyone from record company video executives to producers, directors and costume designers. At the same time,

careful and thorough audio preparations have become essential in every phase of producing Music Videos — from the delivery of the master audio tape to the production company, to the final phases of post-production. At every step along the way, however, it is still common for audio problems to arise. While awareness of the audio engineer's requirements has grown, it is clear the further education is still in order.

Audio Masters for Music Video

Among the discussions held at last Fall's *Billboard* Video Music Conference — perhaps the industry's largest year-end gathering of Music Video professionals — was one panel devoted to various activities of "The Behind-the-Scenes Team." The panel included representatives of virtually every technical craft involved in the production of music videos — including audio engineering. Representing the audio perspective was Gordon Skene, a veteran film and television sound engineer whose Music Video credits include Michael Jackson's *Billie Jean*, plus clips by Toto, Fleetwood Mac, Pat Benatar and countless others.

"Sound appears to be the thing that most people take for granted as just 'being there' in a Music Video," Skene commented in his opening remarks. "But the quality of the sound that ends up on a video clip pretty much depends on the record company and the master tape of the song that it provides."

Elaborating on his remarks after the conference, Skene explained that "the problem with a lot of record companies is that the people in the video departments, who make a lot of important decisions on a production, are not always aware of how *crucial* a first-generation sound source is. We've gotten master tapes that would drive you nuts — 7½ ips quarter-track, highly questionable stock, that sort of thing.

"The other problem I have is with different versions of the song. Say there's a four-minute version and a five-minute version. Maybe the New York office only has a master of the five-minute version, and we're shooting to the four-minute version. We sometimes get stuck with a situation where somebody makes quick dub of something, and that's what we use as our audio master. I did a Tyrone Bronson video for CBS where the New York office accidentally sent us the *wrong* tape. The artist's management was freaking out. The only available recording of the piece was on disc. So we ended up having to transfer from the record!"

No matter where or how it originates, a low-quality master can compromise the effectiveness of a clip before the first day of shooting begins. The less-than-ideal scenario that Skene lays out is confirmed by another audio engineer in the music video field, George Johnsen, proprietor of EFX Systems in Burbank. Johnsen first appeared in "Visual Music Scene" in connection with his resourceful sound effects work on Michael Sembello's "Automatic Man" (see April 1984 issue of *R-e/p*). Over the past year, Johnsen has had the chance to witness his share of problems with audio masters that have come through EFX.

"We sometimes have a problem with the label sending out a third- or fourth-generation copy of the song," he relates. "In post production, we have to go down at least two more generations to get the music back to the video master, so the audio quality gets pretty poor. There are now a number of video production houses though, that are requesting either the original master of the song or a safety copy of the master from which to make their transfers.

"So, in many respects, things are getting better; in other respects they're getting a little worse. The labels are starting to be aware of the fact that a sync tone and timecode are needed on the master, but they're not always putting them on correctly. We sometimes get half-inch copies that, supposedly, have sync and timecode on them, but, unfortunately, the sync [timebase] is not always referenced to the timecode."

Which brings us to the topic of digital music masters for Music Video projects. As the practice of mixing to digital two-track grows, the record labels are beginning to furnish digital masters to video production companies. In addition to improving audio quality, PCM masters recorded on videotape help out with some common sync problems as well. This aspect was pointed out in a follow-up interview with Tom Davis, whose contribution to Philip Bailey's *I Know* (one of the first Music Videos to incorporate dialog and sound effects) and Van Halen's *Jump*, a video on which he tackled the problems of a final music mix that differed from the mix used in shooting the clip, were detailed in the April 1984 "Visual Music Scene."

"You don't run into as many sync problems with digital masters," Davis explains, "because they have their own control track and sync track. Also, because of the very nature of digital audio, you can't play a digital tape off reference. It only takes one reference frequency — which is 59.94 Hz [the NTSC video field rate] — and that's that."

Davis, Skene, and Johnsen all report that they've worked on several projects in recent months for which the record company furnished a digital master, commenting that those projects went more smoothly and sounded better than projects based on analog masters. As the popularity of digital recording and mastering spreads, audio engineers specializing in Music Video will no doubt be working with more and more digital master material. For the time being, they are doing the best they can with what they are given. *. . continued overleaf —*

Location production of a video for "No Way Out," from Jefferson Starship's recent album, *Nuclear Furniture*, directed by Irv Goodnov. Pictured: David Freiberg and Paul Kantner.

CREATIVE CONCEPTS PRODUCES *CATS* MUSIC VIDEO

Creative Concepts, a new Music Video production company, earlier this year completed a five-minute video of "Rum Tum Tugger" from the Broadway production of Andrew Lloyd Weber's musical *Cats*. Principals in the production company are Lou and Richie Vetter, co-owners of New York's Blank Tapes Studios, who have joined forces with Jeff Lee, production stage manager, and T. Michael Reed, *Cats*' dance supervisor.

Featuring actor/singer Terrence V. Mann, who performed the song in the original Broadway cast, and just finished filming a lead roll in the film version of *A Chorus Line*, the $250,000 *Cats* Music Video is described as the first to be produced for a Broadway musical. "Music Videos of feature film and recording artists have already proved their worth as promotional tools, but this is the first time a theatrical production has tapped the medium," says Richie Vetter. *Cats* recently launched a national touring company, and a third production played in Los Angeles earlier this year. The Shubert Organization expects our Music Video to become a powerful asset in spreading awareness for the show."

In addition to its distinction as the first Broadway musical Music Video, "Rum Tug Tugger" is also said to be one of the first videos recorded digitally. Blank Tapes Studios owns a Sony PCM-3324 multitrack, and Vetter considers that digital recording was crucial to maintain the integrity of the original music. "The original cast recording features a large orchestra of more than 2C pieces," he explains. "We felt that a cleaner, more open sound would work better for the video, and incorporated synthesizers and a drum machine when we re-recorded the song at our studio with Terry Mann. We then added guitar and horn tracks. With our digital recorder we were able to eliminate loss of quality due to multiple transfers, and we came out with a sound that was far superior to traditional analog technology."

"As Stereo TV becomes more widely entrenched in the marketplace, the recording techniques for Music Videos are going to have to become far more sophisticated," offers company principal Richie Vetter. "If Broadway musicals and touring shows find that our *Cats* video is having a positive effect on the box office, we expect to see a real flow of production work from that quarter. With our combination of recording expertise and Broadway musical background, we hope to carve a solid nitch for Creative Concepts in this newly emerging area."

The *Cats* video was filmed over a six-day shooting schedule, with three days on stage at New York's Longacre Theatre where a special set was constructed. "It was impossible to shoot at the Winter Garden Theatre where the show was based," Vetter recalls, "because our lighting, sound, and time requirements would have played havoc with the show's performance schedule."

Director of photography was Michael Negrin, cameraman for many Billy Joel videos. David Seeger of Today Video served as editor, cutting feature film-style to pre-recorded music tracks. Co-producer Lou Vetter credits Willie Austin of Glen Glenn Sound, L.A., David Smith of Editel, NY, and Sid Zimet of Audio Force, NY, with providing solid advice on digital recording aspects for the video. "The technology is still quite new," Vetter explains. "Sid Zimet was particularly helpful in synching the digital audio track to the rough cut picture for the final underscore and mix. There weren't many of these Sony digital recorders in the field at the time, and it was very helpful to have experts like these available to answer questions based on their hands-on experience.

"We came to terms with the evolution of the recording industry," he continues, "and decided that if we are going to continue to maintain an active part in the business we *have* to change with it. Our purchase of the Sony PCM-3324 digital recorder, our new Studio D, which was especially designed for audio-for-film and -video, and our entrance into Music Video production with Creative Concepts are all major steps designed to place us in the front ranks of the music and Music Video production scene." □□□

◀ Cast of Andrew Lloyd Weber's *Cats* during production of a five-minute video for "Rum Tum Tugger," which is probably the first Music Video to be recorded digitally using a Sony PCM-3324 multitrack. A special stage set was constructed for the shoot at New York's Longacre Theatre.

Remix of digital multitrack tapes of the *Cats*' "Rum Tum Tugger" at Blank Tapes Studios, New York. L-to-R: Jeff Lee, Lou and Richie Vetter, of Creative Concepts.

Visual Music

After some bad experiences, Skene reports, "the production companies I work with have adopted a more-or-less standard policy for working with audio masters. Whatever we get — hopefully it's the best possible master, but that's not always the case — goes over to Glen Glenn Sound [Hollywood], where it is transferred to a 15 ips half-inch tape. Channels one and two have stereo audio; channels three and four, respectively, have SMPTE timecode and 60 Hz sync tone. That tape then becomes the master tape for the project, from which they they strike a Nagra copy for playback in the field and all other copies that they need. This helps to keep things uniform, but by the time you lay your final audio back to your video master, you're ostensibly three generations down, no matter which way you look at it."

Audio on the Set

As a specialist in on-location audio, Gordon Skene has seen his responsibilities increase over the past year from simply playing back the music track on the set to recording dialog and effects and, on occasion, juggling the problems of recording audio during playback. Although it is a fairly straightforward operation, playback of a track on the set can entail a few difficulties. Because of budgetary considerations, or simple misunderstanding of what is required, Skene explains, the production company will often request a playback PA system that is woefully inadequate for the task at hand. Although the playback sound does not turn up on the final videotape, it can, says Skene, be crucial to an effective performance on a music video.

"Musicians are often incredibly apprehensive about performing in a video, especially if it's their first; they like to hide behind their music a bit. They want a substantial volume level in their playback. American bands, in particular, like to work with performance-level playback and will play live at a level to match that — if they're doing a performance clip. The artist's attitude often is: 'I want to be able to really feel the track so I can give out the kind of intense emotion you want on this film.' If the producer isn't willing to give the band this, chances are he is going to get a lousy performance.

"What I generally use in a situation like that is a standard BGW 750 power amp. I'll also use a graphic equalizer on occasion. A lot of people think that is going too far for just a playback system, but it enables me to get different settings for the two output channels of the BGW. On one channel, for example, I can emphasize high-end for the drummer, and then equalize the second channel to suit the other members of the band. I will sometimes go through a small Yamaha mixing board as well, just to have a little more control over the playback. What I use for monitors depends on the band and the situation — it can be anything from two small stage monitors to four large cabinets like the Klipsch La Scala."

Most dialog recording for Music Videos takes place on the set; looping is generally reserved for big-budget productions. Also, when the lines are to be spoken by the musical artist (i.e., not by professional actors who are familiar with looping procedures), many directors feel they can capture a better performance on the set. The fact that there are usually only a few lines involved makes this approach a practical one.

"The techniques used to record dialog for a Music Video are no different than what they would be for a film," Skene offers, "as far as microphones, mike placement and the aural perspectives involved. Engineers coming from a video or recording studio background might think about it differently, but I just draw on my film background. Generally, I will use a Sennheiser 815 or 816 on a boom, and hire a boom operator. I've also been using some Schoeps microphones lately, and find they have a very natural sound that is good for this kind of work."

As in film-sound recording, radio mikes are planted on actors and musicians that have lines to speak, and are used in long shots where it would be difficult to keep a microphone boom out of the frame. Although this approach works well in most cases, as Skene relates, minor difficulties sometimes arise.

"This is an area where the wardrobe department gets involved. For example, I recently did a video with Apollonia [Prince's sultry musical cohort] where we decided to use radio mikes. We found, though, that it was physically impossible to use a radio mike on this girl because she wasn't wearing enough clothes to conceal one! We were shooting exteriors in a park and had to do the best we could with a shotgun mike. Luckily, it ended up working out fine."

In the majority of cases, there is no need to have the music track playing back while dialog or effects are being recorded on the set. Once in a while though, it becomes necessary for some or all of the people on the set to hear the track while recording is taking place — in order to mime instrumental or vocal parts, for example, move in time to the music, show a reaction to the music, etc. In such cases, the people who need to hear the track are outfitted with earpieces containing miniature receivers. For performance videos, audience reaction scenes that require audio recording on the set are often handled with the aid of a "boom box."

"It's a low-frequency tone generator that functions as a metronome," Skene explains. "We figure out the tempo of the song, and the tone generator will put out regular pulses at about 30 or 35 Hz. It's barely audible. During the mix, we can just EQ it out."

Post-Production Audio

The problems of incorporating dialog and effects into Music Videos also extend to post production. Facilities like George Johnsen's EFX, which combine record experience with post-production work for film and TV, have emerged as problem solvers for directors working in this new genre. Over the past year, EFX has provided transfer and sweetening services (including dialog and effects work) for video clips by Molly Hatchett, Earth, Wind and Fire, Los Lobos, Ratt, Chaka Khan, and Diana Ross, among others. According to Johnsen, the ability to integrate record-quality audio technology with film/TV techniques is what has made EFX a popu-

lar post-production house for music videos.

"What you need is a facility with TV/film experience, but which is geared to *music*," he comments. "Obviously, you can't have a Music Video with dialog that sounds like standard television fare; it wouldn't go over on a high-fidelity music clip. You have to match up your music, dialog and effects, making sure that the dialog is clean and that you don't have nasal-sounding dialog tracks encroaching on lead vocals. The tricky part of it was making sure that the dialog that was recorded [on location] matches the environment created in the visual part of the program."

But by far the most frequently cited audio/video integration problem encountered — and one that has cropped up a good many times in these pages — is the problem of effectively synchronizing sound with picture. In most cases, the difficulties have their origin in the more-or-less standard practice of shooting music clips on film (be it 35mm or 16mm) and then transferring to videotape either for editing or audio layback. Problems arise from the fact that film equipment runs at a speed which is referenced to a resolve frequency of 60 Hz, while video equipment is referenced to 59.94.

Johnsen explains how problems of this nature can arise, and details EFX's procedure for making the necessary speed conversions:

"We start by making a half-inch, four-track transfer of the song, which is provided by the record company. The four-track has timecode and a 60 Hz reference tone printed on it to provide us with a constant sync reference — the original audio master goes back to the record company and is never used again.

"We then make Nagra field copies from our half-inch four-track. The Nagra has a feature called 'pilot tone,' which is the 60 Hz crystal-generated — or crystal injected — pilot frequency that allows you to have that machine resolve in playback. In other words, it plays back at *exactly* the same speed at which it was operating when it recorded [the guide music track]. The film cameras are also crystal controlled to a reference of 60 Hz, or a subdivision of 60 Hz, so that they are running at *exactly* 24 fps or 30 fps.

"When you go to post," he continues, "99 times out of 100 you're transferring [film to videotape] on a Rank Cintel [telecine], which is referenced to 59.94 Hz. Since your shooting reference was the equivalent of 60 Hz [which corresponds to a film rate of either 24 or 30 fps] there is a speed shift from 60 Hz to 59.94 Hz. We therefore have to speed correct our half-inch, four-track audio master so that its 60 Hz reference conforms with the 59.94 Hz reference of the Rank Cintel.

"We start with the 60 Hz that was on our half-inch and reference that to 59.94 Hz, so that the audio is slowed down [by a corresonding amount]. Then we lock up our half-inch machine to a one-inch, C-format VTR and transfer the audio tracks over to videotape. This means that we now have a one-inch, C-format editing master that is speed corrected to match the Rank Cintel master."

The speed-conversion problem can be worked out in a number of ways, according to Tom Davis, who also reports seeing film/tape problems plague a good many Music Video projects. "An audio tape can have either 60 Hz or 59.94 Hz [sync/resolve tone] on it. It can have just timecode, or timecode plus one of the two reference frequencies. The main thing is to be *consistent* in what you do throughout the project. That is why I *strongly* recommend that a qualified audio supervisor should be hired — either by the record company or the video production company — to oversee the project. I've seen *too* many producers and production companies try to handle it themselves. They will have a rental company supply the audio equipment, a different company do the transfers . . . too many unrelated people get involved. No one knows what the others did, so you end up with a tangle of different formats in post production."

According to George Johnsen, film/tape speed conversion problems have also served in the past year to inhibit the practice of remixing the music to suit the visual perspectives of the finished picture.

"Only a couple of clips have been remixed that way this year," he reports. "Artistically, remixing for the video is definitely the way to go, but if you remix the song to match the video, that means you have to speed-correct the 24-track master. Usually, you don't find a 24-track master with enough open tracks to accommodate the necessary reference code and timecode."

Digital Technology

The practice of digitizing the final audio mix that is laid back to the video master is one that has gained popularity during post production on Music Videos.

"I'm seeing a lot more digital audio going on," Tom Davis observes. "I won't say it's being used on every clip, but it is happening on a regular basis. On a number of projects I've done that were specifically designed for home video distribution on LaserDisc or Beta Hi-Fi, we delivered audio on either a one-inch or ¾-inch videotape master that had a soundtrack digitally encoded with a Sony PCM-1610. There are still high-quality audio projects where we don't use digital, of course. What we often do in those cases is lay back the audio to the one-inch videotape master with Dolby A noise reduction, and then they will decode it in the mastering process.

"A lot of times we'll deliver a four-track audio tape master with Dolby-encoded stereo tracks and the same timecode as the video master. A four-track audio tape machine will give you much better quality than a one-inch video machine. But delivering digital audio is better still. That's true even if all prior stages of the project were analog. If you digitize the final audio mixdown, the resulting audio tracks will be extremely quiet, and you won't suffer generation loss."

Davis feels that the popularity of digitizing the final audio for Music Videos will continue to grow. Many engineers, he explains, are still a little unsure of the correct procedures for digitizing audio tracks, but these doubts are likely to diminish with time.

"A lot of people are getting hip to the fact that there's no big trick to doing a digital master," says Davis, "as long as your engineering staff is used to it. The 1610 format [for example] will work on any helical videotape machine. Typically, we use the ¾-inch Sony BVU-800 U-Matic Series at Pacific Video [the facility at which Davis is based], but you can even go to the one-inch format. You basically just lock up two machines and decode the audio from the playback machine, through the 1610, onto the record machine."

The process *does* demand careful attention to a few details, however. For one thing, Davis stresses, it is important to remember to reference the digital sampling rate of the 1610 to 59.94 Hz.

"Plus," the engineer adds, "you have to remember that the 1610 essentially becomes the timebase corrector for the [video] machine that is recording, which means that you have to disconnect the [VCR's] timebase corrector, if it has one. Typically, if you rent a 1610 from a rental house, they send you the 1610 with a ¾-inch videotape machine. You plug them together and it all works. In an editing house you have to be careful though, since all the machines are usually set up for editing, and have their own timebase correctors, which are interfaced into the facility's computer system. So you basically have to strip down the timebase corrector on whatever machine you are using, and let the 1610 become the timebase corrector."

Looking Ahead

Digital audio, as Tom Davis points out, no doubt will play a role in solving the two most glaring audio problems that still face Music Video — those of overall audio quality, and synchronization between the various visual and aural elements of a program. But the real key to solving both problems lies with the people involved in Music Video as much as the technology. Improved lines of communication between those responsible for every phase of a Music Video project is the only real means of achieving the type of consistency that insures high-quality audio and freedom from sync problems.

At the present time, Music Video provduction is still very much a haphazard business. The popularity of the genre has exploded much too rapidly and unpredictably, it can be argued, for the industry to form a detailed and uniform approach to creating visual music. This facet, however, is the very thing that has made Music Video so fascinating to watch in the past few years — both on the screen and behind the scenes. It shares this quality with all the Big Moments in pop music: the birth of rock in the Fifties; the British Invasion of the Sixties; the Punk/New Wave onslaught of the Seventies — they've all been *enfants terribles*, whose alarming rate of growth far outstripped parental control.

But, having made it out of infancy, Music Video must necessarily face the organizational strictures of adult life. Last year the first steps were made toward unionizing the Music Video industry. It's too early to say how unionization would affect the audio aspects of the genre — or, for that matter, its overall artistic content and quality. Meanwhile, audio engineers themselves have felt a need for better organization of their craft within the Music Video industry, and have come up with several suggestions of their own.

"Maybe there needs to be an RIAA-type of organization for Music Video," Gordon Skene offers. "An organization which insures that everyone is given a certain set of guidelines for tape formats, procedures, and things of that nature. Or maybe it could take the form of an AES pamphlet (or series of pamphlets) explaining the technology, such as Nagra sync and other film procedures, to AES members. Sound for films is *definitely* an area for which audio engineering schools should offer a few classes."

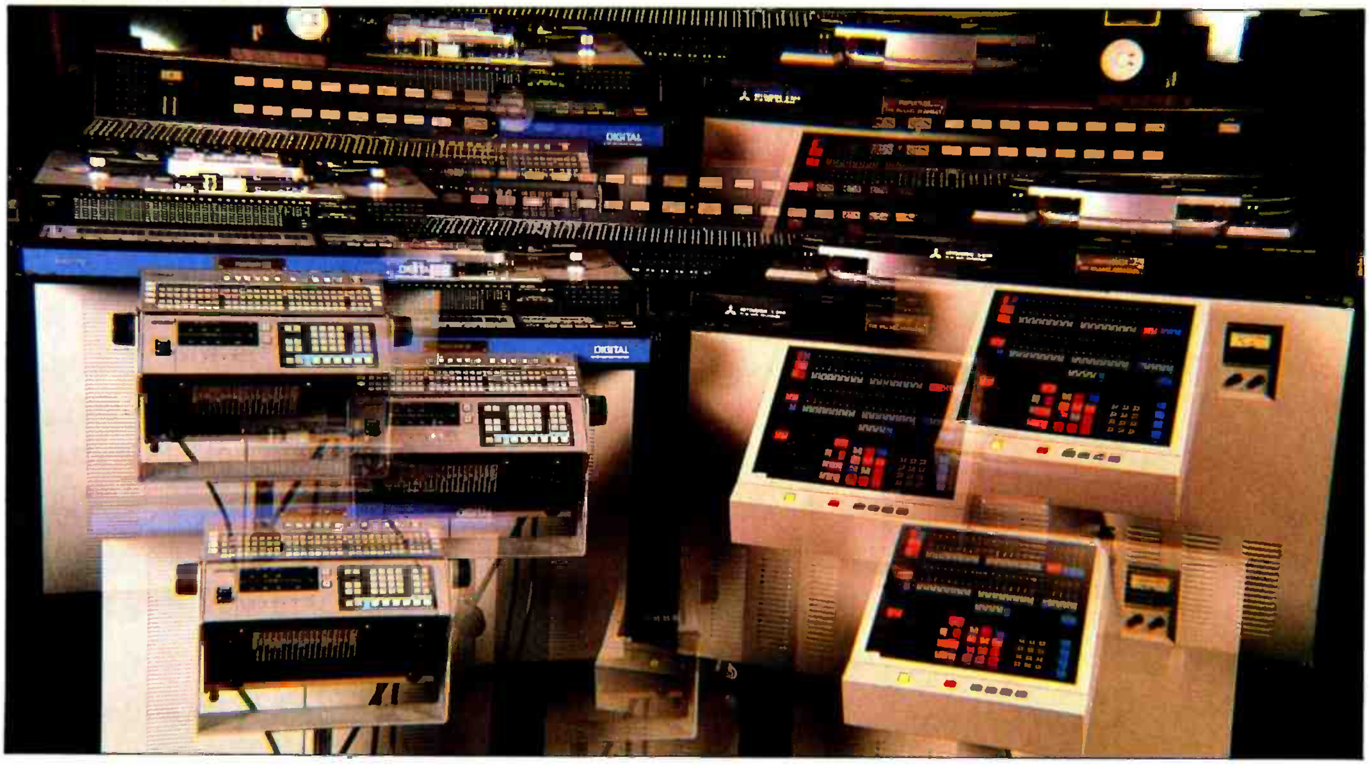
A central organizing body responsible for technical standards is an idea that holds a lot of appeal for many audio engineers working in the Music Video field. Visual music, though, may be still too much of a maverick industry to accept this type of organization.

"You have a lot of real individuals who are doing Music Video," comments George Johnsen, "and they're coming to it from all different kinds of backgrounds. In putting together any kind of central organization, the first problem you would have is an attitude of 'This is *my* magic way of doing it, and I'm not going to change it for you.' Secondly, a lot of people involved in the industry are far too busy to participate in this kind of organization. Thirdly, Music Video is just not an organizable type of industry right now, because of the type of creative people who are involved in it."

On a more grass roots level, several facilities, including EFX, have begun holding informal, free seminars for Music Video professionals that want to learn more about the audio aspects of the craft.

"Our seminars are usually about 20 people strong," says Johnsen, "so we're able to deal with specific examples and do the sort of things that would be impractical at a larger industry gathering. We're dealing with a new medium, and we think we can help people get a little hipper to what's involved."

Education is also what the "Visual Music Scene" has been all about. Throughout recent columns, I've attempted to focus on the most significant aspects of Audio-for-Music Video, and will continue to do so in the future. The coming year promises to be a pivotal, "stand-or-fall" period for visual music — aesthetically, commercially, and technologically. Stay tuned. ■■■



SONY PCM-3324 and MITSUBISHI X-800 digital multitracks at Studio D, The Village Recorder, Los Angeles. Photography: Elizabeth Annas

SYNCHRONIZING DIGITAL MASTERING AND MULTITRACK SYSTEMS WITH ANALOG TAPE, FILM AND VIDEO TRANSPORTS

An Overview of the Interface Capabilities of Sony, Mitsubishi, Studer, 3M, JVC and EIAJ-Format Systems

DIGITAL TRANSPORTS: Why are they so Different from Analog Machines?

an introductory article by Rodney Pearson, service coordinator, Solid State Logic, Inc.

Just when you thought it was safe to go into the studio, having learnt all there was to know about synchronizers, SMPTE timecode, and even the dreaded Drop Frame, along came digital tape machines with new buzz words such as Sampling Frequencies, Error Correction and Word Clocks. This article will present an overview of the methods of synchronizing digital audio recorders to analog recorders, video recorders and film transports, with particular reference to new techniques for film and video production.

Now that the major audible advantages of digital recording have been (more or less) accepted, some of the secondary advantages can be addressed. The manner in which digital audio is recorded requires the 16-bit data stream be timed by a system clock to which all digital processors, such as analog-to-digital (A/D) converters, must by synchronized. This system clock offers several advantages: it allows, for example, absolute synchronization of audio tracks with other audio or video transports, while simultaneously eliminating speed variations and wow and flutter.

Digital audio is similar in many respects to video; in fact, several systems actually record the digital data on conventional video recorders. Even when synchronizing machines with SMPTE timecode, a timing signal at the Sampling Frequency — known as the Word Clock — must be transferred from the master machine, so that not only will the tapes run in sync, but all the processors will be synchronized to one another.

If digital data is to be transferred between machines it is essential, of course, that *all* tapes are recorded in the same data and track format, and at the same sampling frequency.

There are several important differences in synchronizing digital recorders, compared to analog machines. For example, the system clock runs at a frequency of 4 or 5 MHz, and supplies timing information not only to the digital processors, but also to the transport capstan and reel motors (or drum servo in the case of VTR-based machines). So, it is not sufficient to simply input a 9,600 Hz waveform to the capstan motor and thus control its speed until sync is achieved. Several synchronizers actually replace the system clock with a voltage-controlled oscillator (VCO), and thus vary both the processor *and* capstan speeds to maintain sync. Because of the precision of these timing circuits, analog recorders should always be "resolved" if they are to be used as a master transport for slaving to digital or video recorders. ("Resolving" simply means that a machine's timecode is itself locked to a stable time base before additional machines are slaved to it.)

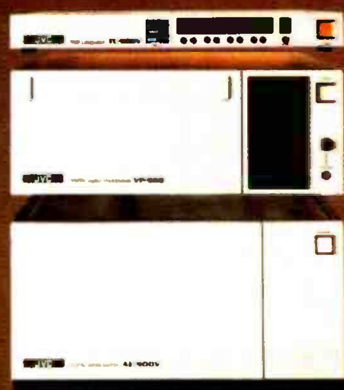
. . . continued overleaf —

JVC Digital Audio. The artist's editing system.

Digital audio editing takes on new speed, simplicity, and flexibility with JVC's 900 Mastering System. Anyone with a trained ear can learn to operate it in minutes and be assured of professional results of outstanding fidelity, accuracy, and clarity. And while sonic excellence is surely the 900's most persuasive feature, flexibility runs a close second; for not only will the 900 operate with ¾" VCR's, but with VHS cassettes, too, with total safety and confidence, making it ideal for mastering digital audio discs and the increasingly popular hi-fi video discs.

The DAS-900 consists of four principal components.

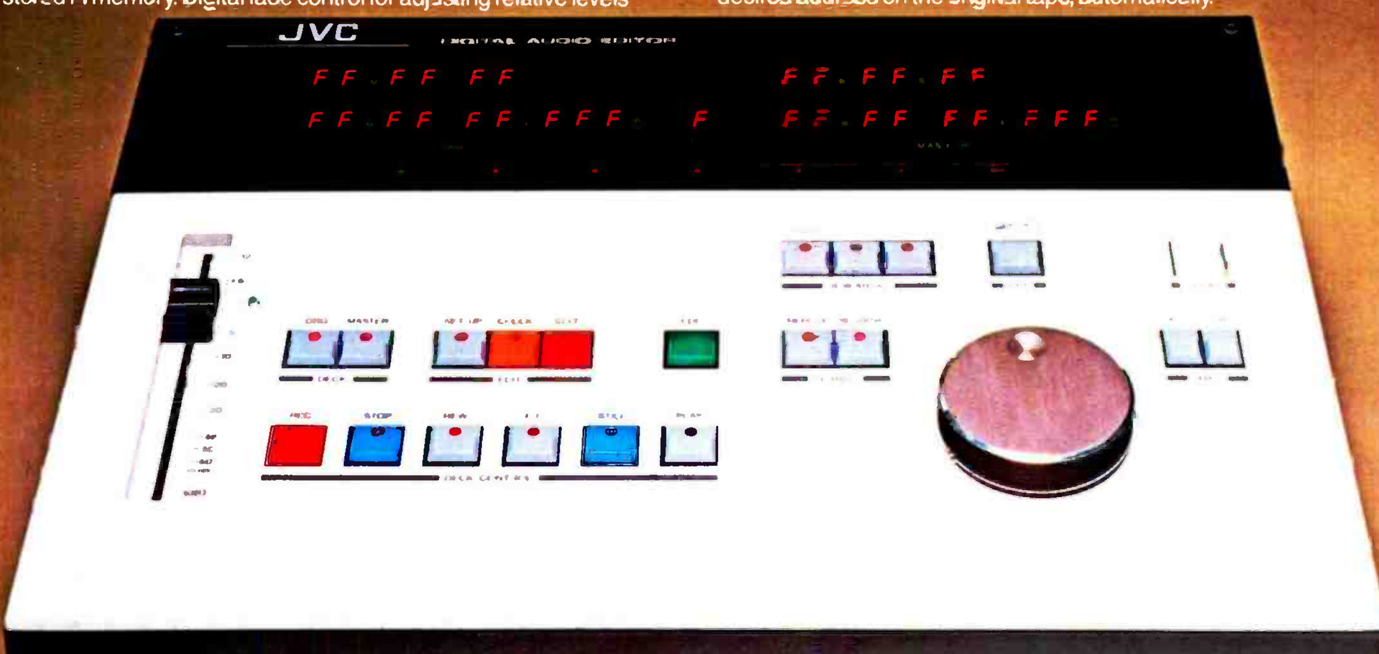
VP-900 Digital Audio Processor. Two-channel pulse count mode processor. Several 16-bit microprocessors make it compatible with other professional production equipment such as cutting lathes, synchronizers, and encoders. Dynamic range of more than 90 dB. Frequency response from 10 to 20,000 Hz (± 0.5 dB), and low recording bit rate of 3.087 Mb ts/s at 44.1 kHz. Transformer-less analog I/O circuits further improve sound quality and the analog-to-digital, digital-to-analog converter reduces distortion to less than 0.02 per cent, while an emphasis circuit improves signal-to-noise ratio. Logic circuit uses CMOS LSI chips for high reliability, compactness, light weight (48.6 lbs) and low power consumption.



Audio Editor Control Unit. Electronic governor for routing, coordinating, and executing all edit functions, both automatic and manual. All commands, from digital dubbing of original to master for continuous programs, to repetitive point-to-point manual cueing are regulated here.

TC-900V Time Code Unit. Actually two time code units in one, this unit reads and generates SMPTE standard time code and synchronizes the JVC exclusive BP (bi-parity) time code. Thus the DAS-900 will operate effectively with both time codes; a necessity when the System is to be synchronized with video equipment.

AE-900V Digital Audio Editor. Simplicity itself to operate, this little number puts editing right in the hands of the artist, if need be. Precise to within microsecond accuracy, edit search can be carried out by manual cueing, automatic scan, or direct address. It will confirm cut-in, cut-out points independently by recalling signals stored in memory. Digital fade control for adjusting relative levels between original and master tape. Shift function for changing edit points backward or forward in 2-ms steps for super-fine adjustment. And variable-gradient cross-fading function for smooth continuity at the edit point, variable in 0, 10, 20, and 40 microsecond steps. Auto tape locate function enables the user to locate the desired address on the original tape, automatically.



For a demonstration of the DAS-900 Digital Audio System, a Spec Sheet, or JVC's complete catalogue, call, toll-free

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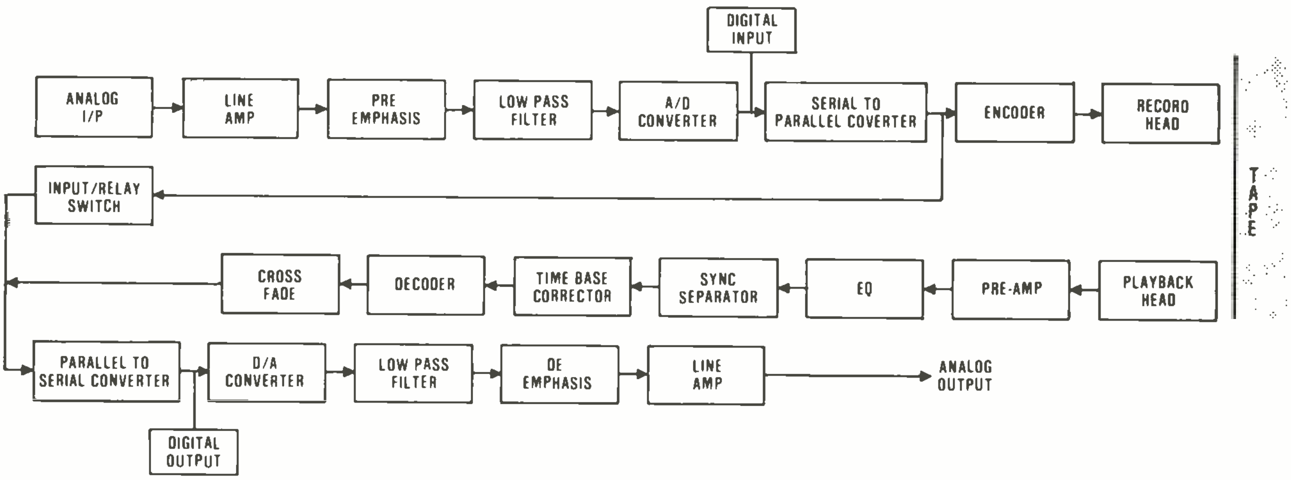

FIGURE 1: SIMPLIFIED BLOCK DIAGRAM OF A "TYPICAL" DIGITAL RECORDER

DIGITAL PRODUCTION

The nature of digital audio, with this precise timing and lack of generation loss, means that a great deal of manipulation of tracks is possible — for example, there are systems in which it is possible to synchronize up to 15 digital multitracks through their respective remote control units. Since this adds up to a total of 360 digital audio tracks, any number of which may be transferred without degradation from one tape to another, the possibilities for those who are considering joining the Procrastinator's Association (I know several producers who are seriously thinking about becoming members!) are literally endless. (Assuming, of course, that you have a console with 360 inputs and the budget for almost $2 million worth of tape machines!)

But seriously, this ability to "bounce" tracks without degradation is one of the major reasons for using digital audio instead of analog in film or video recording.

One of the reasons often cited by film-sound engineers for continuing to use large numbers of mag film machines, instead of analog multitracks, for dubbing is the simplicity with which individual tracks may be "slipped" — that is advanced or retarded to maintain sync with the visual action. Working with mag, tracks may be slipped by as little as one sprocket hole (a quarter of a 24 fps frame, or about 10 milliseconds).

Using digital multitrack recorders, however, two methods of slipping tracks are available. Individual tracks may be retarded with millisecond resolution by simply inserting a digital delay line in the audio path. Another method, if time permits, is to copy the required track or tracks to another machine that is synchronized to the master via timecode. Then, by manipulating timecode offsets, tracks may be advanced or retarded. If tracks are transferred digitally, there will be no audible degradation or generation loss. The recent unveiling of digital effects processors and Winchester- or Hard Disk-based edit-

. . . continued on page 76 —

SYNCHRONIZATION AND EDITING FUNCTIONS OF THE STUDER D820X-2 DIGITAL RECORDER

by David Walstra, product specialist, PCM Systems, Studer International

The extensive synchronizing and editing capabilities of the D820X-2 Twin-DASH format digital two-track are made possible in large part by an advanced transport-control system. The transport's spooling and DC capstan motors are all under software control, allowing microprocessors to calculate and control tape movement in the fastest and most efficient manner. Spooling speeds as high as 15 meters per second (590 ips) permit very rapid access to cue points. Acceleration and deceleration is smooth and positive, even with 14-inch tape reels. (The 14-inch reel capacity allows the D820X-2 to record over two hours at 15 ips.)

The D820X-2 has extensive locate control facilities for fast access to the audio program. Locate keys are also under software control, thus allowing individual units to be programmed for specific customer requirements. The transport may be locked to external time references, such as word clock, sector clock, or composite video sync signals.

The varispeed control allows for a deviation of ±12.5%. Of course, the digital output signals will vary accordingly, but a Studer SFC-16 sampling rate converter can be utilized to provide automatic correction during transfer to another digital recorder.

Remote Control and Synchronizing

Following the DASH format, the D820X-2 provides a separate timecode channel; all versions of the machine will have a timecode reader located in the deck with meter bridge display. The deck provides both serial and parallel remote control facilities. Parallel remotes allow for simple remote connections, and also provide required signals from the tape deck to the remote, while the serial (RS232, RS422) bus allows for easy interfacing to synchronizers, such as the Studer TLS4000. (Both the D820X-2 and TLS4000 will be compatible with the forthcoming SMPTE/EBU serial bus format.)

Editing Capabilities

The D820X-2 is designed to facilitate both tape-cut and electronic editing. Tape-cut editing is possible with complete confidence because of the DASH-Twin implementation, which offers data redundancy and "smart" data protection. Electronic digital audio editing will be accommodated with a system to be introduced by Studer following final market introduction of the recorder.

Even without an electronic system, the D820X-2 permits compilation of master tapes by digital-to-digital transfer. Levels can be adjusted in the digital domain during copying, and the timecode channel ensures accurate starting and stopping times.

Cue channels are provided for monitoring in the spooling and cueing modes.

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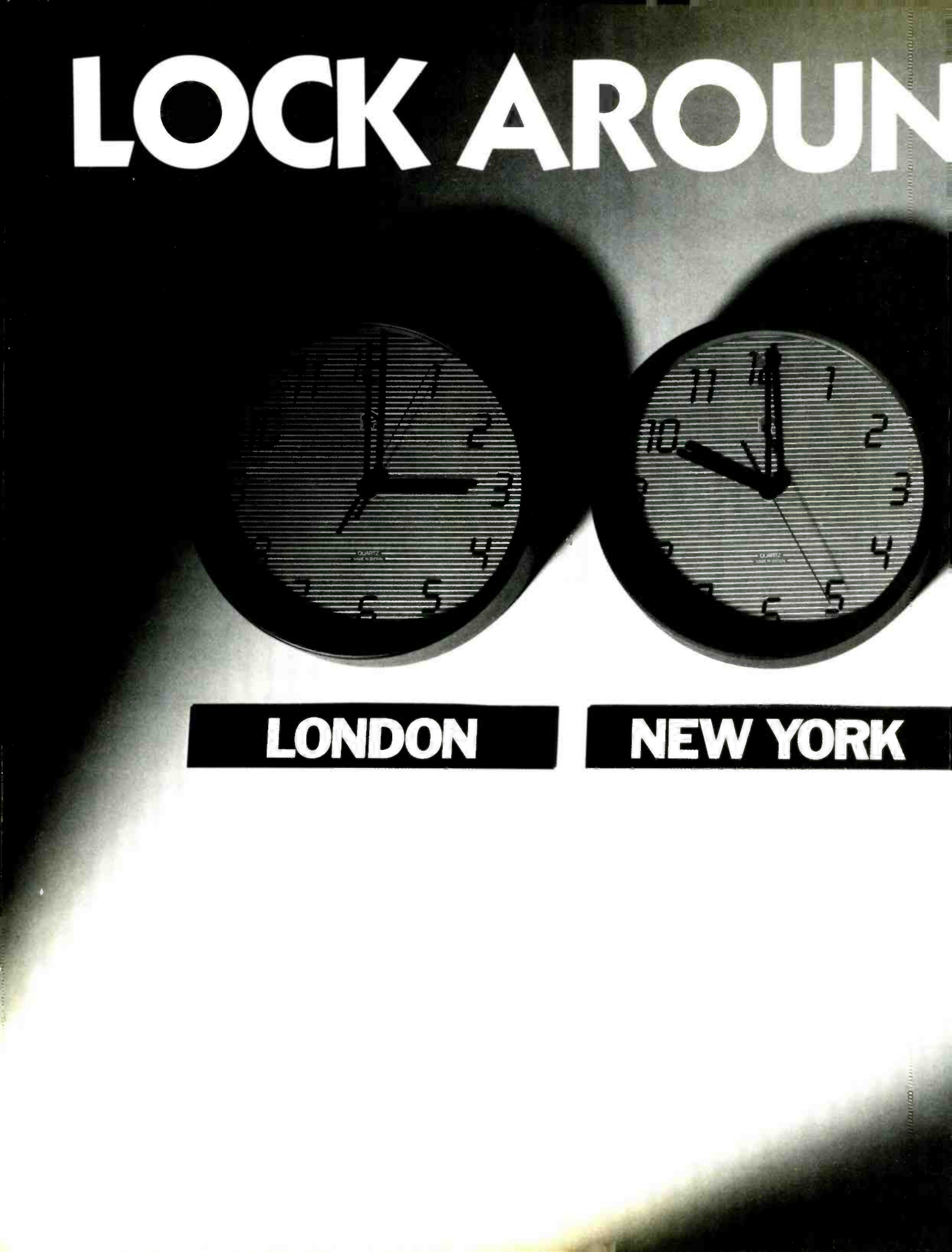
▼STUDER D820X-2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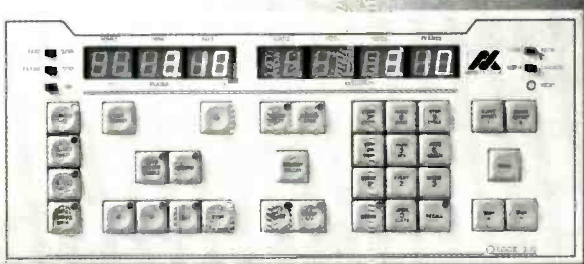

The ability to synchronise video and audio recorders is an increasingly vital facility required in studios all over the world. As much as three-quarters of today's audio recordings involve a visual aspect and recording is more international than ever before. Basic tracks in New York, string and brass overdubs in London, dubbing in Los Angeles . . . modern international productions need an international standard for machine synchronisation, and there's really only one: Q-Lock by Audio Kinetics.

It's the same all over the world. The simple, uncluttered controls. The custom interfaces that suit your machines. The remarkable software capability. The integrated system with built-in expansion possibilities. It all adds up to accuracy and ease of use, and that means speed and creative flexibility. It means Q-Lock. If you're looking for an international standard, you've found it.

Now, more than ever, you need to keep in sync with the times. Lock around the clock – with Q-Lock.

Audio Kinetics Inc.,
4721 Laurel Canyon Boulevard,
Suite 209, North Hollywood,
California 91607, USA
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AUDIO KINETICS

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DIGITAL PRODUCTION

ing systems, such as the Compusonics DSP-2000 and AMS Audio File — both of which are expected to offer direct digital inputs and outputs compatible with current recording formats — will allow such devices to be used without any loss of signal quality.

Digital Multitrack Operation

At this point, a brief description of how a typical digital multitrack recorder works might help to dispel any fears you may have about this new technology. So, if you pay attention, we'll learn about sampling frequencies, bit rates and error correction, but I promise we *won't* get involved in Nyquist Theorems or Sampling Algorithms, or any such advanced theory.

As shown in Figure 1, the analog input signal is fed to the line amp and level control, after which optional pre-emphasis may be added to optimize the signal-to-noise ratio. The 20 kHz lowpass filter prevents *Aliasing* by eliminating frequencies greater than half the *Sampling Frequency*. The analog signal is converted to digital, and the resultant *Serial Data* stream converted to *Parallel Data*. It is then encoded (in which data is *Interleaved* and *Parity Bits* added) and recorded by the record head. (See the accompanying Glossary for a detailed definition of these concepts.)

On playback, the signal is amplified and equalized before passing to the sync separator, which performs data extraction, demodulation and Cyclic Redundancy Checking, marking any errors with "flags." The serial data is then converted to parallel data, and any jitter in the playback signal absorbed by the Time Base Corrector, before the Word Data and *Error Correction* flags pass from the TBC to the Decoder, which *De-Interleaves* word data and performs error correction. The Cross Fader is used during Punch-In and Punch-Out sequences, crossfading between input and playback signals. Following the Input/Playback switch, Parallel data is converted to Serial data which, in turn, is then converted from digital to analog, lowpass filtered, de-emphasized (if required) and passed to the output amplifier.

As can be seen from the above explanation, data is being moved and processed at extremely high speed; our present sampling frequencies of 44.1 or 48 kHz are only the starting point. Thus it is imperative that *all* processors are locked to a common clock.

To see a practical example of the amount of data required to accurately reproduce a piece of music, consider the following example: a Compact Disc has a capacity of 70 minutes of music, yet if the same disc is used to store computer data, its capacity is a staggering 600 megabytes — more than 3,500 times that of a floppy disk used on a Commodore 64! A single CD ROM, as they will be called, would be capable of storing the entire *Encyclopedia Britannica* or over 1,000 books, each with 300 pages containing 300 words; and still have space available for graphics to be displayed on a TV screen.

Towards the Future

The long and often complicated process of transferring sound from location recording to theatrical release may involve at least five or six generations, including ¼-inch analog and 35mm formats. The resulting loss of transient response and degradation of signal-to-noise ratio has been endured by anyone that has paid good money to enjoy superb visual images accompanied by dubious audio reproduction.

The transfer of audio to videodisc, Beta or VHS Hi-Fi for replay in a high-quality home video/audio system may involve two more generations of one-inch Type-C video before reproduction on a system with audio specifications superior to those of any single link in the transfer process. In every one of the transfer stages (other than copying to Beta or VHS Hi-Fi), the audio quality suffers much more than that of the picture.

Since it is now possible to record digital audio on location using porta-

SYNCHRONIZING 3M DIGITAL MULTITRACKS WITH FILM AND VIDEO TRANSPORTS

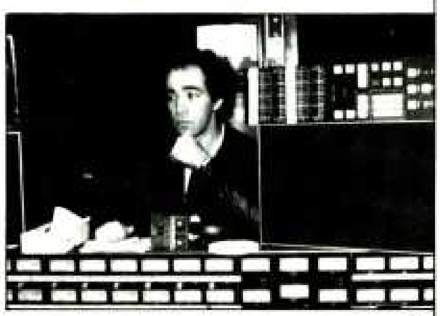
by Frank R. Dickinson, president, Digital by Dickinson

During the past few years, there has been considerable interest in the locking of digital audio transports with video and film chains. As this interest slowly grew, this writer began conversations with several of the controller and synchronizer manufacturers in the hope that we would be able to buy a complete system. Since we could not find such a system, we set out to develop one that could do the job.

We felt that for frame-accurate editing, there were two basic considerations: one, to find a controller that would allow the most flexibility, ease of operation, and a system configuration that would be friendly to operate; and, secondly, to devise a means of interfacing the 3M Digital Mastering System with each other, to allow the transfer of digital data under control of a SMPTE timecode editor. And it was only after considerable investigation that this design philosophy was developed. I felt that the controller should synchronize the two machines, lock them together and transfer the data digitally. The controller would be used to coordinate the master and slave machines and find the correct in/out record points, thereby creating the ability to generate a digitally transferred tape equivalent to the Edit Decision List (EDL) supplied by a video editor.

The controller selected to handle this task was the Audio Kinetics Q.Lock, since A-K was the first company to develop an interface for 3M DMS digital system. At this point, with sketch pad and trusty wire-wrap tool in hand, an interface was constructed that allowed the Q.Lock to bring the two machines into synchronization at the correct SMPTE timecode location, and then to transfer the data digitally from the slave machine to the master machine according to the EDL supplied. Since the 3M machine is presently configured to allow the transfer of just one or all of the 32 available tracks, this same control capability is passed on to the timecode editing system. Such a fine degree of

▲ **A pair of 3M DMS 32-track machines aboard the Record Plant Mobile during an audio/video shoot for the *Stevie Wonder Comes Home* Showtime special.**



▲ **To provide a more accurate indication of recording levels during the *Stevie Wonder Comes Home* shoot, the DMS remote control and level-display unit was mounted on top of the Record Plant's API console meter bridge, in view of the engineer.**

ble processors and video recorders, then to transfer and edit this audio in the digital domain without any generation loss, the inherent advantages of digital over traditional techniques are obvious.

For those wishing to take advantage of digital recording without disrupting established post-production routines, digital audio may be recorded on location and transferred to analog 35mm mag film for conventional editing by film editors. If this film is encoded with SMPTE timecode, an Edit Decision List (EDL) can then be generated, allowing the original digital audio material to be "conformed" by an editing system to duplicate the 35mm edits.

As the techniques used in recording studios merge with those of video and film Post-Production, new capabilities will be developed. As each of these disciplines learns from the others, the full potential of digital audio synchronized to other media will be realized.

□□□

SYNCHRONIZING SONY PCM-1610, PCM-3102/3202 AND PMC-3324 DIGITAL SYSTEMS

by Curtis Chan, senior engineering manager, Sony Corporation of America

The increasing prevalence of digital equipment in recording and post production studios makes it critical for today's engineers to possess a thorough understanding of new interfacing and synchronization techniques available today. And there can be little doubt that digital technology is having a growing impact on production techniques. Recently, Neil Young completed a Music Video using two 24-track digital recorders. Glen Glenn Sound has produced a 30-minute, demonstration film that features a six-channel, digitally mastered soundtrack. And *Metropolis*, Fritz Lang's 1926 classic film, has been enhanced by a new digital soundtrack produced by Giorgio Moroder. While digital technology formed the common bond for these diverse projects, the actual recording procedure was as different as their respective subject matter.

Synchronized Recording

The basic analog recording configuration includes a master recording unit linked to one, two or more slaves via a timecode synchronizer. When using a Sony PCM-3324 digital multitrack as the master recording unit, and a Sony PCM-3102/3202 digital two-track for the slave, the slave recorders are phased locked by using external word-clock connections. Also, the sampling rate of the slave recorder should be set to the same frequency as the master unit. The same synchronization technique applies on occasions for which PCM-3324 is set-up as the slave unit; refer to Figures 1A and 1B for additional details.

The PCM-3324 can also be expanded to synchronize with video recorders. An optional accessory board, the DABK-3000, enables the recorder to lock to an NTSC monochrome signal at 30 Hz, NTSC color at 29.97 Hz, and even PAL at 25 Hz. Additionally, 24-50-and 60-Hz reference signals can be selected. Use of the DABK-3000 Video Clock Board allows audio and video editing to be synchronized through the use of a common sync source (composite video). The versatility of using a videotape recorder as the master can also be realized, composite sync being fed to the synchronizer as well as to the VTR, PCM-3324 multitrack and PCM-3102/3202 digital two-tracks. (The configuration is shown in Figures 2A and 2B.)

The use of several multitracks in sync makes it mandatory to incorporate a master remote control unit. The Sony RM-3310 Remote Controller enables up to three PCM-3324s to be operated in perfect synchronization. The controller also features an instant and precise autolocator with offset capability, plus simple, single-cable connections to the recorders. In order to set up a 72-channel system, three RM-3310 controllers must be used with three PCM-3324 24-tracks.

When hooking up recorders in tandem, synchronization accuracy is always a prime concern. The use of one IF-3310 interface with the RM-3310 and two IF-3311 mixing console interfaces with their respective RM-3310, guarantees that the three machines will lock-up to within a one-bit synchronization accuracy. Up to 72 channels of console Rec/Ready control will also be possible, as shown in Figure 3. The SRIF-1 is a parallel

SYNCHRONIZING 3M SYSTEMS — *continued . . .*

control at times provides the ability to carry notes or small sections of material across edit points, allowing smoother edit transitions than are possible by conventional butt editing.

The first major use of this in-house system came during the postproduction editing of "Stevie Wonder Comes Home" special for Showtime. [See feature article in October 1984 issue of *R-e/p* for further details —*Editor.*] By the time the show was aired via satellite, the audio tracks had been assembled using the 3M digital editing system, mixed via analog transfer to another two-track digital system, sweetened using audience tracks from a second digital machine, and then transferred to a video-based PCM digital system for broadcast. (This same system is currently being used to assemble the audio for a soon to be released Ricky Scaggs' live show that was recorded in London.)

The 3M digital 32-track is also presently in use in Nashville with a JVC VP-900 two-track PCM processor to strip and replace the audio for a video promo. In this particular application, the only requirements are that the loss of audio quality be halted through the use of digital recording to remove and process the audio. Such an approach would prevent further loss of audio quality while the video is being edited into the format required by the video clip service. The introduction of digital processing has greatly increased the quality of the video's audio, and has become quite useful in the light of the fact that an increasing number of videos are actually being released in VHS and Betamax HiFi formats, both of which can take advantage of high quality audio masters.

During the time it has taken to develop our system, the increased need for automation has made it necessary to design interfaces for both Solid State Logic and Neve consoles. In the case of SSL, interface design was relatively simple, requiring some minor buffering of the data lines. In the case of Neve consoles equipped with NECAM fader automation, the problems became more interesting, however. The solution was to use an Adams Smith timecode reader and linear translator. This approach allowed the system to generate timecode synthetically from tach pulses derived from the 3M transport, thereby providing the NECAM system with continuous timecode positions that it requires for matching its data to that of the tape.

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Figure 1A: Sony PCM33-24 and PCM-3102/3202 slaved to 3324 master. (VClock boards not installed in transports.)

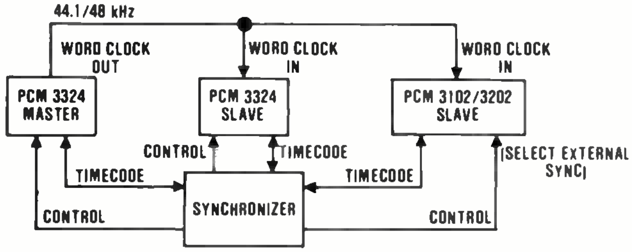
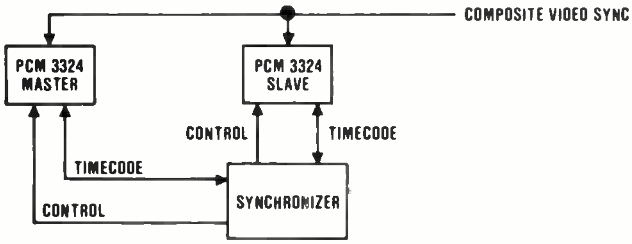


Figure 1B: As Figure 1A, but with VClock boards fitted to PCM-3324 multitrack transports.



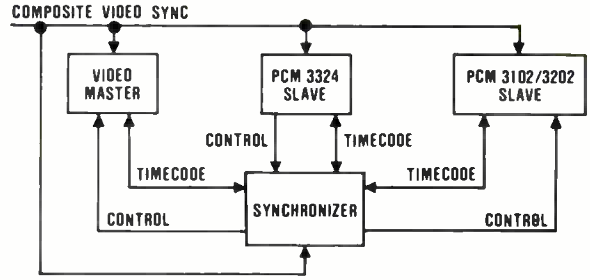
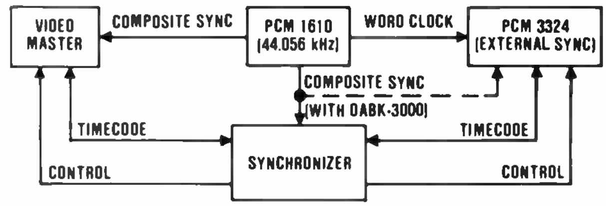


Figure 2A (*above*): Conventional Digital Audio/Video synchronization setup.

Figure 2B (*below*) Digital Audio/Video synchronization using PCM-1610 to provide sync reference for PCM-3324.



interface with standard command/tallies common to all synchronizers. The SRIF-4 applies to the Rec/Ready control on each channel of a multichannel recorder performed by a mixing console, and includes tallies for Rec/Ready/Safe status and Rec On/Rec Off Status.

Under certain circumstances fast lock-up time must be achieved and, as shown in Figure 4, the analog source is the master. The resolver is fed composite video sync and receives timecode reference from the PCM-3324. During play modes, the resolver locks to incoming sync and outputs a corresponding reference signal to the phase input of the multitrack. The advantage of utilizing a resolver in this set up is that lock-up time is minimized, so that sound output occurs almost instantly. In addition, an analog recorder can be used as a master, which is quite useful during sweetening sessions.

Digital Production for Film and Video

Today, many projects are being shot on film, edited on video, and then finally assembled on film, a method that offers the improved speed and ease of video editing. During post production of *Kenny Loggins Alive*, a cable television project and later videodisk release, a PCM-3324 and PCM-1610 were used for optimum sound recording, editing and mastering. So successful was the sound of the final videotape, the production has been used to demonstrate the powerful effect that could be obtained by using a digital dub made from an analog master. [See feature interview with engineer/producer Bruce Botnick, published in the February 1983 issue of *R-e/p* for further details —*Editor.*]

Produced before the introduction of the digital PCM-3324 recorder, the Loggins project made use of an edited video master striped with timecode, which served as a visual post-production reference while the 24-track analog master was mixed down to two-track digital via a PCM-1610 processor. Crossfade audio editing was accomplished with the Sony DAE-1100 Digital Audio Editor, a technique that allowed imperceptible audio transitions between sections, even when the production called for songs to be added from different performances.

First-generation audio for video and videodisc masters was accomplished by synchronizing the two-channel digital mix with the video

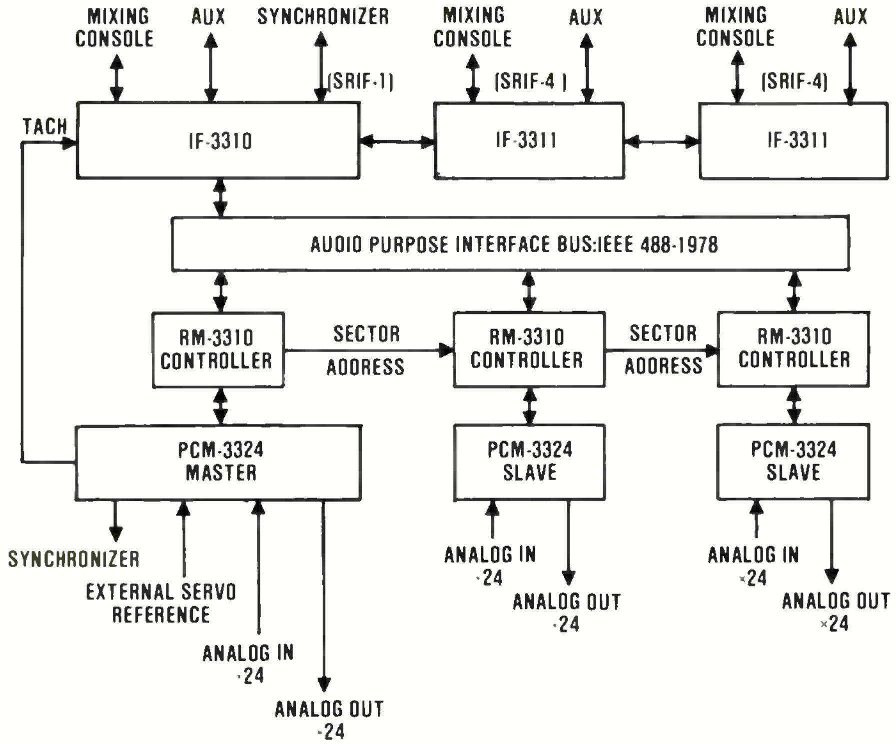


Figure 3: Synchronizing three PCM-3324 multitracks to provide 72-track capability.

master, and dubbing both audio and video simultaneously.

Kenny Loggins Alive was shown at a recent NAB Convention by using a 24-track digital dub made from the analog master. To provide a quality comparison between analog and digital sweetening, for the presentation the one-inch video master was locked to a PCM-3324, a Sony analog 24-track, a PCM-1610 and a Sony two-channel video playback recorder, via an Audio Kinetics Q.Lock synchronization system, as shown in Figure 4B.

The transfer of images from film to videotape can be done in either of two ways: using Telecine or a pre-striped dummy timecode reel that is sprocket locked to the film chain — see Figures 5 and 6 for further details. When using a converter, such as a Rank-Cintel flying-spot scanner Telecine and an external timecode generator, both the converter and the VTR are locked to the same composite videosync reference being fed to the timecode generator. For the second method, the timecode dummy reel must have its start mark matched to that of the optical print, and started simultaneously to achieve sprocket interlock. While video and audio are dubbed over by the converter, timecode is recorded on either the digital multitrack's timecode or one of its analog tracks. When scenes are edited, the 24-track tape can be offset the required amount, and then the timecode resynchronized before re-dubbing.

Synchronizing a film projector to a separate digital multitrack recorder for playback in a theatre, or for final dubbing on the re-recording stage, can be readily accomplished, as depicted in Figures 7 and 8. Since the PCM-3324's DABK-3000 board can lock to 60 Hz reference, both the projector and the multitrack can receive the same reference frequency. The film projector is interlocked with the dummy timecode reel, and the PCM-3324 will have the same reference time as that of the dummy. A Magna-Tech 9F, a unit that converts film footage and frames into standard timecode information, can also be used. If the digital multitrack cannot be locked to an external sync source, then a PCM-1610 can be used, the

GLOSSARY OF DIGITAL AUDIO TERMS

•**Sampling Frequency** is the rate at which the Analog Voltage produced by audio is *Sampled* for conversion to Digital bits or numbers. Common frequencies are: 48 kHz for professional multitracks; 44.1 kHz for Compact Disc and video-based systems; and 44.056 for video referenced to a 59.95-Hz timebase.

This sampling frequency must be at least double the highest audio frequency to be recorded; otherwise *Aliasing* will occur. (In other words, any waveform must be sampled at least *twice* per cycle; otherwise it cannot be accurately re-constructed upon playback.)

•**Anti-Aliasing Filters** are very steep lowpass filters operating above 20 kHz that eliminate any frequencies within the input signal that could cause *Aliasing* problems, as described above.

•**Serial Data** is a method of transferring data in which digital bits "march in single file," as opposed to *Parallel Data* in which they "march" side by side eight or 16 bits at a time. (Note that parallel data requires a separate address line for each of the eight or 16 bits, hence we get 8- or 16-bit microprocessors at the heart of most computer systems.)

•**Word Clock** is the timing reference for recording data. Digital bits are grouped to form digital words, which in turn are synchronized by the word clock running at the sampling frequency.

•**Interleaving** is a method by which adjacent digital words are shuffled before recording so that, should any drop-outs occur or data be lost, the *Error Correction* circuiting will be able to reconstruct the missing data and *De-Interleave* it. (A modern-day example of not putting all your eggs in one basket!)

•**Parity Bits** are added to the data so that the total number of bits in a particular group or Digital Word always add up to either an odd or even number — determined by whether *Odd* or *Even Parity* is being used.

•**CRC** or **Cyclic Redundancy Checking** is the most powerful form of *Error Correction*, and is performed by adding extra bits to the data following certain rules. If these same rules are observed and checked on playback, any errors caused by dropouts and other system nasties can be detected and corrected. (In a 16-bit system, for example, the detection probability using CRC is 99.9985%.) □□□

Sony PCM-3324 digital 24-track

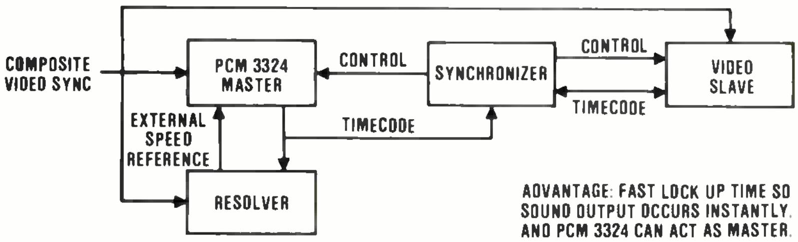
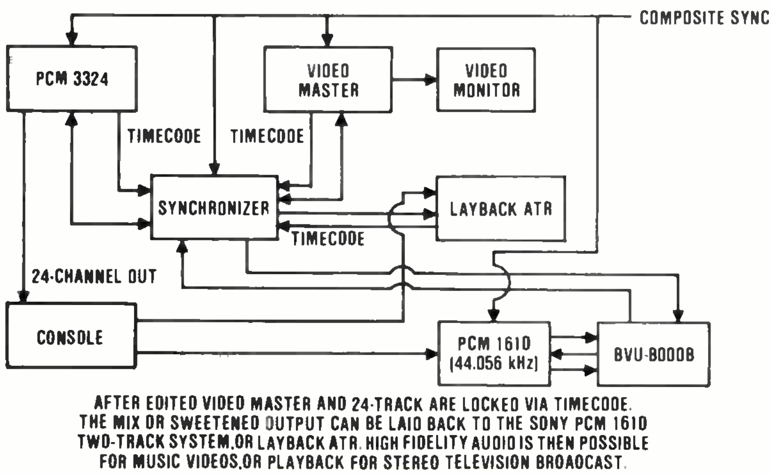


Figure 4A (*above*): Digital Audio/Video synchronization configuration utilizing a resolver to provide improved system lock-up times.
Figure 4B (*below*): Audio/Video layback and sweetening configuration, with video master slaving an analog video-layback transport, a PCM-1610 processor, and a PCM-3324 digital multitrack.



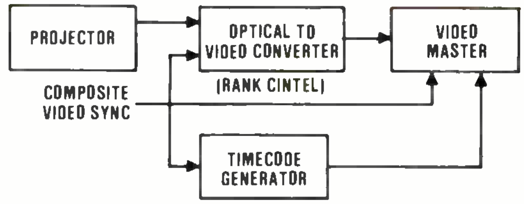
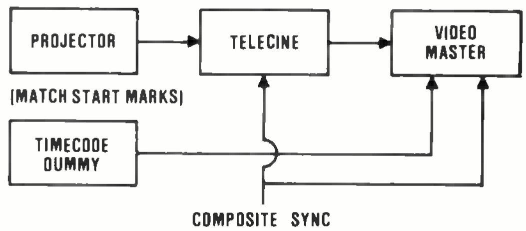


Figure 5 (*above*): System configuration for film-to-video transfer, with timecode generation.
Figure 6 (*below*): Alternate system configuration for film-to-video transfer, using a sprocket-interlocked 35mm dummy loaded with pre-recorded timecode.



tracks being routed via a console for the final mixdown to a Dolby Stereo matrixed, Lt-Rt mix.

Last year, Glen Glenn Sound dramatically demonstrated the numerous benefits that digital recording holds for the film industry. *Digital Dream* proved that even a digitally-recorded master, when transferred to analog for distribution, provides superior sound quality. A digital six-channel soundtrack was prepared during re-recording of the 30-minute film, and then mixed to Lt-Rt for transfer to 35mm Dolby Stereo prints. During music scoring, two Sony PCM-3324 digital multitracks were synchronized with a BTX (now Cipher Digital) unit to provide 48-track capability at Glen Glenn/Record Plant Studio M.

Dialog and sound effects from the field were recorded on a Sony PCM-F1 system and laid back to digital multitrack in the studio. Sony PCM-3324s were then interlocked to picture via the Glen Glenn PAP (Post Audio Processing) system of computerized synchronization. In this way, pre-mixes of hundreds of separate tracks were recorded on the digital multitracks, preventing generation loss during the multiple transfer stages. Later, predubs were combined for the final six-track digital mix, and then dubbed directly to the 70mm print for distribution. . . . *continued overleaf—*

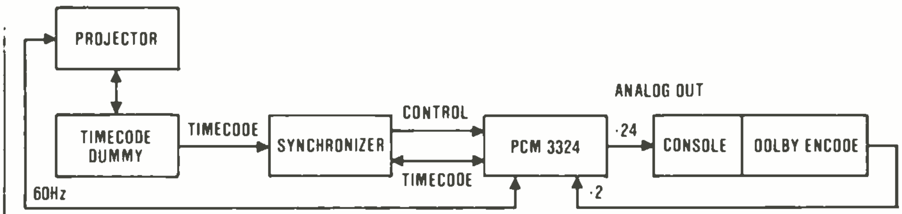


Figure 7 (*above*): Film interlock using DABK-3000 video clock board and 60 Hz reference during re-recording of Dolby Stereo Lt-Rt mix to PCM-3324.

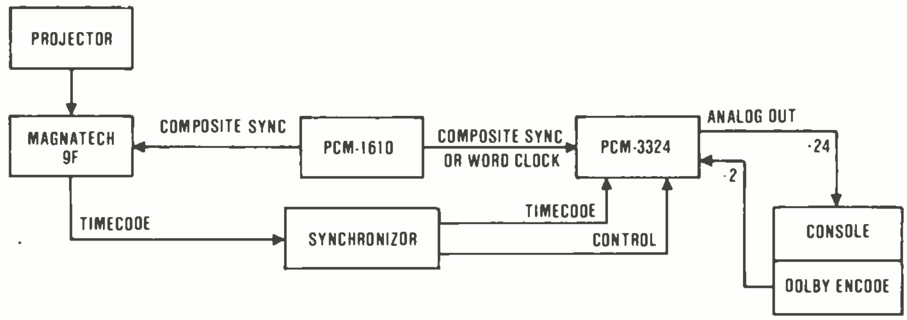


Figure 8: Film interlock system using the PCM-1610 processor to provide composite sync signal and word clock for PCM-3324.

The marriage of digital audio and visual programming has a great future, thanks to the dual-impact of Music Video productions and the introduction of Stereo Television. Dramatic improvements in sound reproduction were recently brought to life by several Music Videos. The final Dolby Stereo two-track master of The Talking Heads' *Stop Making Sense* was transferred directly to the optical negative from a final Lt-Rt mix recorded on a PCM-3324 digital multitrack. A pristine signal quality was maintained throughout the re-recording and mixing of the program by the use of digital equipment.

Stevie Wonder Comes Home was a recent Showtime television special that used two-track digital audio from a PCM-1610. The simulcast production was synchronized and uplinked via a Scientific Atlanta converter system, and transmitted to Showtime affiliates in digital form via the Satcom 1R satellite. [See the October 1984 issue of *R-e/p* for full details of audio post production of the *Stop Making Sense* and *Stevie Wonder Comes Home* presentations — *Editor*.]

Still another post-production method was used for Neil Young's Music Video, directed by Hal Ashby, with Elliot Mazer producing audio. The original multitrack masters were mixed using two PCM-3324s running in sync with the picture, and a PCM-1610 two-channel system used to receive the mix prior to final layback to the video master.

Movie audiences as well as television viewers are now enjoying digitally recorded sound. Giorgio Moroder's remake of Fritz Lang's 1926 classic film, *Metropolis*, was one of the first major motion picture released to rely on digital technology for the soundtrack. Three PCM-3324s were cascaded, and the 24-track music masters mixed over to four tracks of alternate PCM-3324s. These mixes were then run in sync with a video workprint of the film until the music exactly fitted the film's newly edited version. A PCM-1610 provided word sync to the digital multitrack, plus video sync to the BVU-800 used for video playback. An Audio Kinetics Q.Lock synchronizer was locked to the video sync output of the PCM-1610, with timecode running at the video rate. For theatrical distribution, music was crossfaded from the dual multitracks and recorded directly to the four-track Dolby Stereo film master.

The re-premiere of *Metropolis* was shown to audiences at the Academy of Motion Picture Arts and Sciences, Los Angeles, with audio being replayed from twin PCM-3324s interlocked via SMPTE timecode to the film.

From this first project, audio professionals quickly recognized that digital is now quickly replacing the traditional methods of 35mm film recording; for example, it is very easy to synchronize two PCM-1610 processors to replace the familiar four-track mag-film recording methods. The music scores of *E.T.*, *Poltergeist*, *Indiana Jones and the Temple of Doom*, plus *Explorers*, recorded by Bruce Botnick, were all digitally mastered using this method, with the soundtrack albums being mastered from a four-channel mix.

The use of digital is not limited to studio use either. Frank Zappa recently videotaped and digitally recorded a live concert in New York City with his personal-use PCM-3324 synchronized to VTRs via composite sync. Tracks were then mixed directly onto the PCM-3324 by combining and bouncing the composites onto two free tracks for the stereo mix. Final mixes for Zappa's *Does Humor Belong in Music* were then transferred through the PCM-1610 (44.056 KHz sampling frequency) at Pacific Video, Hollywood.

The young world of digital recording is quickly uniting with the video and film worlds to provide dramatic sensations never before possible. *Metropolis* more than typifies this progress: an innovative movie in 1926, the film is again a unique production almost 60 years later thanks to the use of digital technology.

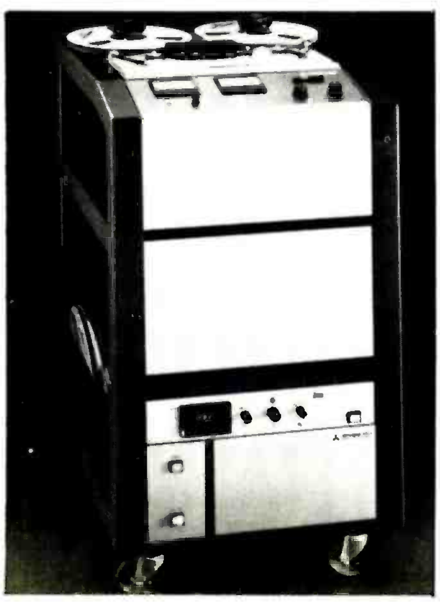
Unlimited possibilities exist because of the flexibility digital technology possesses. And recording professionals will continue to find innovative ways to increase the pleasure we receive from sound produced digitally. □□□

SYNCHRONIZATION OF MITSUBISHI X-800/850 AND X-80 DIGITAL TRANSPORTS

by Cary B. Fischer, operations manager, Mitsubishi Pro-Audio Group

Synchronization of the Mitsubishi digital 32-Track and X-80 digital two-track machines is both standard and common, as compared with the interface of the analog tape machines presently in use in studios today. Both units are supplied with a parallel interface for external control of all transport and electronic functions. The "universal" RS-422 (RS-232C) differential serial port also is provided on the new X-850 cut-and-splice digital 32-track; this additional interface will accommodate the requirements of video editing suites, television studios, and any other operations that base their tape-machine control protocol through serial interface. Other features available on the new X-850 include the addition of improved external clock interface capabilities, which can be set to either 9.6 or 8 kHz. The user can lock to external reference, such as 50 or 60 Hz, 59.94, or composite video — the X-850 will automatically sense the external reference, and then lock to it.

Mitsubishi X-80 digital two-track, with optional VCO-80 synchronizer interface, and DDL-1 disk-mastering delay unit.



To design and build an interface synchronizer cable, computer interface for console automation, or film chain interface, you need to understand what the two prospective pieces of hardware will supply in the way of control outputs and inputs, tally commands, servo information, power, ground configuration, tach information, and direction. (Again, this is standard interface information similar to that found on any professional tape machine.)

One large advantage that the end user gains from utilizing Mitsubishi digital machines is the supply of additional tracks for the purpose of data storage. Both the 32-track and two-track are provided with an additional SMPTE timecode track, which alleviates the problem of using up a valuable tape track for code. In addition to the timecode track, the 32-track has an additional two tracks for analog recording, and two digital tracks for the encoding of automation information, or for any other data that you might need to store in real time. The two-track has an additional analog track, as well as the SMPTE code track, the former being used to cue the tape prior to razor-blade editing.

Setting the damping factor is one overlooked function that's usually an option provided for you by various synchronizer manufacturers. The term "damping," in this context, refers to the use of a synchronizer's capabilities to control the speed, direction, and acceleration of a tape transport. What this means is that you can match the internal control of the transport, via synchronizer control, by adjusting the rate at which the synchronizer changes states to ensure proper tape handling. If proper control of the tape transport is overlooked, you could damage the tape. The X-800 has a software program stored in ROM (read-only memory) that ensures proper tape handling, and which will prevent the tape from being thrashed around the transport by external control of the unit's servo.

The X-80 two-track also features a standard interface to the outside world for master control. If the X-80 needs to be slaved to a second machine, the DEC-VCO (voltage-controlled oscillator) interface will be required. The VCO is an interactive control unit that allows for various synchronizers to control the X-80 servo-control circuitry, and allows for three modes: a) Internal Calibration; b) synchronizer control; and c) Varispeed Operation. The unit's readout displays varispeed information (in percentage), and sampling frequency.

Shown in Figure 1 are the pin outs for the X-80's remote control interface.

Interfacing the Mitsubishi line of digital recorders currently in use at various types of facilities around the world was accomplished with great ease. As Figure 1 shows, there are no special considerations to the interface or synchronizing the machines.

Outside of the familiar use of tape machines interlocked for video scoring, dual-machine operation, and automation machine control, there is an

SIGNAL NAME	PINOUTS	SIGNAL NAME	PINOUTS	SIGNAL NAME	PINOUTS
INPUT					
Ext. Sig	34	EDIT	38	02	10
SLW Wind	35	EXT	1	C-REC	11
STP	19	CNT RST	2	FF	12
FF	36	PLAY	3	GND (+30V)	17
REW	20	REC	4	+30V	33
C-REC	37	FWD	5	0	50
BACK	21	ERASE	6		
P-EDIT	22	SPEED 1/2	7		
T.START	39	OUTPUT			
EXT.MUTE	23				
SPEED 1/4	40	CNT-INH	43		
SPEED 1/8	24	UP/DWN	27		

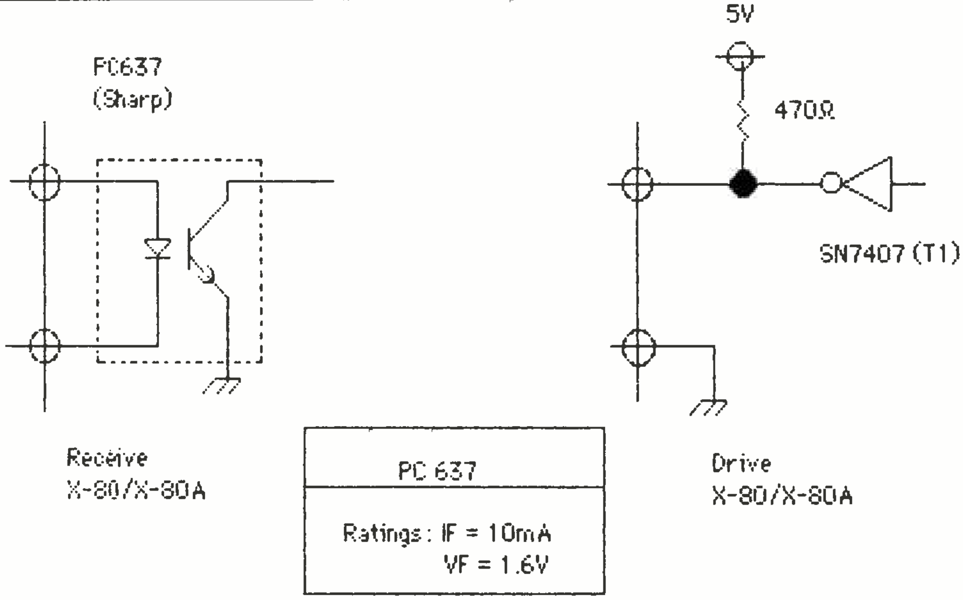
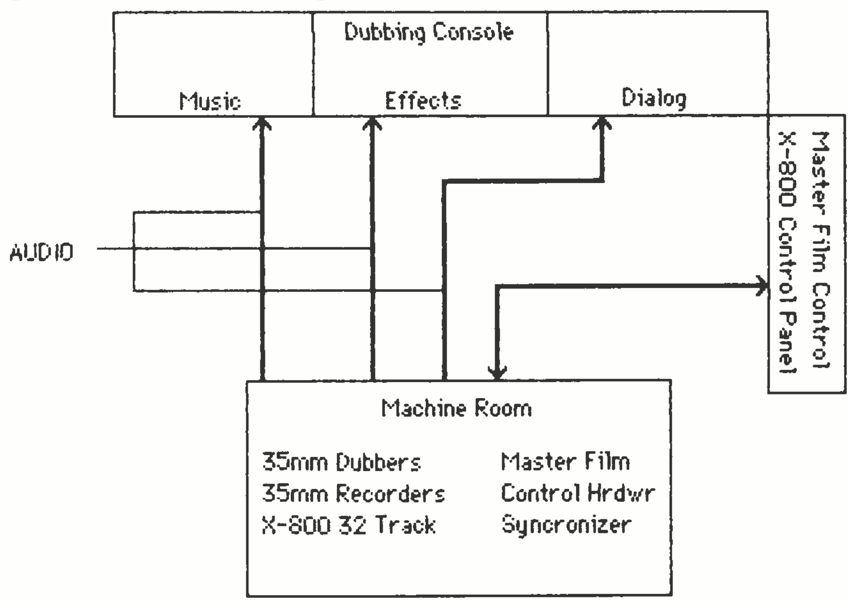


Figure 1: Pinouts for Mitsubishi X-80 digital two-track's remote control interface.
Key: CNT = Counter Reste; C-REC = Code Record Control; BACK = Set back-speed mode for reverese spooling; P-EDIT = Output dubbing data to line-out; O2 = Counter roller output for play mode.

Figure 2: Schematic of film-to-digital-audio synchronization for scoring sessions.



The restyled Mitsubishi X-850 digital 32-track now provides cut-and-spice editing and improved external clock interface.

interesting application now in progress utilizing the 32-track recorder.

TIMECODE: THE KEY TO AUDIO/VIDEO AND FILM SYNCHRONIZATION

The key ingredients to successful interfacing and synchronization in video, film or record projects is timecode, which provides a common "language" that enables various recording equipment to communicate with one another, and gives audio professionals more freedom to create even the most complex recordings. The SMPTE/EBU timecode format adopted in the late Sixties serves as the basis for today's sophisticated video and audio editing and synchronization systems. Two versions of the code currently exist; of the pair — Longitudinal Time Code (LTC) and Vertical Interval Time Code (VITC) — we are concerned with the former. With Longitudinal Time Code, the following relationships apply:

- NTSC 2,400 bits per second: 30 frames per second = 80 bits per frame.
- PAL and SECAM 2,000 bits per second: 25 frames per second = 80 bits per frame.

While each bit has a specific value, of special interest here are the 10th and 11th bits. The 10th bit is the drop-frame bit, and indicates a drop or non-drop frame timecode condition. The reason for this indicator bit being necessary is that NTSC color signals have an actual frequency of approximately 29.97 frames per second, which means that a generator counting at the 30-frame rate would produce an increasing error of 3.6 seconds per hour. To compensate for this error, 108 frames are eliminated from the timecode each hour, or two frames each minute with the exception of the 10th minute. Bit 11 is the color-frame bit, and only applies to color recordings. In NTSC video, the color frame is divided into four fields, each being 1/15th of a second in duration. Fields 1 and 3 are defined as color frame A, while fields 2 and 4 are defined as color frame B. Color-frame identification indicates that even-frame numbers coincide with color frame A, while odd frame numbers coincide with color frame B, an identification that assists video-editing systems in maintaining the correct video signal color-burst phase relationship across the edit points. □□□

18-BIT PCM TECHNOLOGY BEING DEVELOPED BY dbx

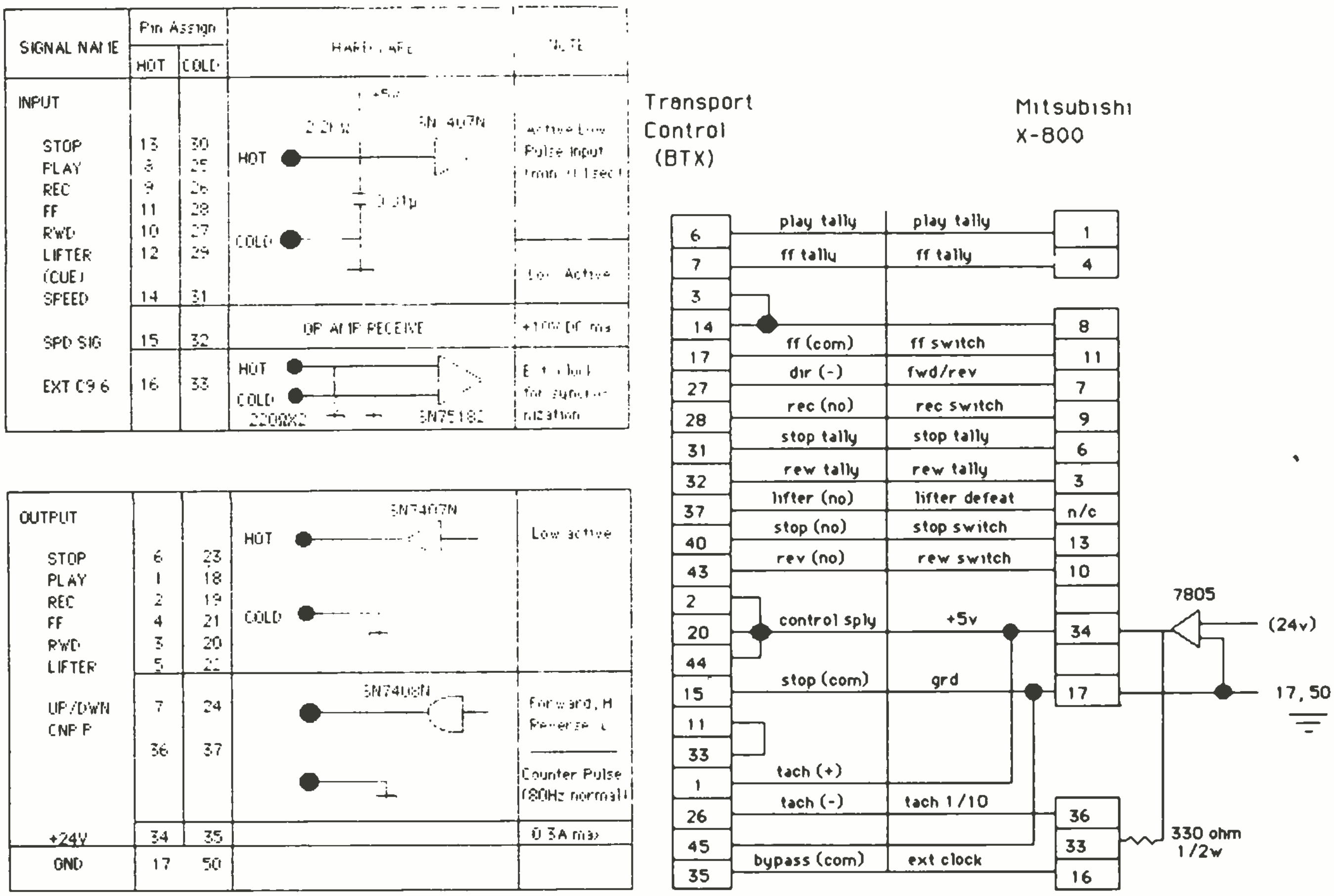
While the main emphasis of these articles is on synchronizing PCM (pulse-code modulation) transports, it should not be overlooked that dbx has available the Model 700 two-channel digital processor based on Companded Predictive Delta Modulation (CPDM) technolgy, for use with ½- and ¾-inch VCRs. While the company has yet to release a purpose-built editor, existing timecode-based video editors can be used for frame-accurate digital edits.

Also under development is a series of 18-bit PCM ICs that the company plans to offer on an OEM basis to manufacturers of tape transports and digital processors. According to Bob Adams, dbx director of research, the new chips make use of a hybrid front-end topography that samples at 6 MHz, followed by a switchable 16/18-bit stage clocked at 48 kHz. The ICs should be available next year. □□□

The Burbank Studios, Los Angeles, has interfaced an X-800 in one of its dubbing rooms, where the unit is being used as a dubbing machine. The X-800 is synchronized using a BTX (now Cipher Digital) Shadow Softtouch to follow film pulse information supplied from a master film controller. The X-800 chase locks to the film transport operation, and acts as though it were a conventional 35mm film dubber — the only difference being the supply of 32 digital audio tracks! The TBS technical department has manufactured its own version of the autolocator, which is being integrated into the console for ease of machine control.

TBS is also using another X-800 in its scoring stage. Apart from scoring music tracks directly to this machine, TBS is also adding additional information to the tracks — in the form of pre-dubs — for use when the tape is brought across to the dubbing stage. This mode of operation using the X-800 is now starting to take hold. Fantasy Studios, San Francisco, also is using the X-800 for film scoring, and a project is underway that will give them the same type of flexibility achieved by TBS, and diagrammed in Figure 2.

It must be kept in mind, however, that slaving a digital recorder to analog, video and film transports, is



INTERCONNECTS BETWEEN MITSUBISHI X-800 AND CIPHER BTX SHADOW

not the easiest task to accomplish from the manufacturer's point of view. Because the PCM code is time based, it takes quite a lot of number crunching in order to reproduce a PCM signal while the transport is responding to external sync control. With the sync signal (located on each track of the tape) comparing its timing to the machine's internal clock, and then being buffered to eliminate wow and flutter, you are really taking a perfect environment — digital data recorded at a crystal-controlled sampling frequency — and turning it into an imperfect environment, in which the data has to be decoded at a variable sampling frequency that matches the external sync perturbations. Despite these admittedly minor limitations, there is little doubt that digital interlock is becoming an increasingly important technique in audio production.

□□□

THE EIAJ-FORMAT DIGITAL PROCESSOR COMES OF AGE

A Review of Available Hardware, Transfer and Editing Systems

by David Smith, Editel/New York

In early 1983, the Sony EIAJ-Format PCM-F1 digital audio processor was introduced to the world via a series of black and white advertisements that centered around a photograph of a man recording a baby grand piano in his home with a pair of microphones plugged into a small grey box. With a domestic videocassette recorder, the ad explained, the promise of high-quality digital audio in the home could become a reality. And indeed it did, although the popularity of such digital processors quickly developed well beyond recording your home baby grand piano.

During the last three years much water has flowed beneath the digital bridge regarding the eventual fate of EIAJ-Format processors — in addition to the introduction of a virtual plethora of accessories, a disussion of which will form the basis for this article. (However, I will refrain from commenting on the future availability of any of these units, since it would tend only to do a further disservice to an information-hungry recording industry. Except, that is, to comment that a recently divulged "legend" has it that the PCM-F1 was never seriously considered as a viable pro-audio product; merely a demonstration of Sony's technological prowess. The unit's commercial success and the implications of that success so surprised its creators, the legend goes, that Sony's response has been mired by intercorporate strife and a long history of confusing disinformation. Not to mention the fact that the company also markets a fully-professional 16-bit audio processor, the PCM-1610, and DAE-1100 editing system.)

To date, several major manufacturers, including Sony, Nakamichi, Sansui, JVC, and Technics, have marketed EIAJ-Format digital audio processors. All these units are basically similar, in that a digital recording made on one will definitely play back on another; trivial differences between processors from different companies simply reflect the varied approaches that each manufacturer takes toward the marketplace requirements.

Sony, due in no small part to its reputation as a high-end audio component manufacturer, currently has three processors in its collection — the PCM-F1, PCM-501 and PCM-701 — all of which are the only devices capable of 16-bit quantization, a factor that endows them, many users consider, with superior audio capability. (It should be remembered that the EIAJ-Format only defines a stereo processor design for 14-bit quantization at a sampling freqency of 44.1 kHz, and the data format used to record the signal on videotape; the Sony processors feature both 14-bit sampling, and a switchable 16-bit mode that "steals" two bits from the error-detection data stream.)

The PCM-F1 is a portable unit that can be operated either from AC power or batteries, and contains integral mike pre-amps; it was originally introduced with a companion portable VCR, the SL-2000. The PCM-701 is a non-portable version of the PCM-F1, without mike pre-amps, and is designed to blend in with a component hi-fi system. The unit is very similar to the F1 in most respects, with the minor exception of some subtle changes in the digital-to-analog conversion circuitry. Both the F1 and the 701 make use of two analog-to-digital converters, and a single digital-to-analog converter that is time-shared between the left and right channels.

Sony's most recent introduction, the PCM-501, incorporates new, dedicated integrated circuits that have substantially reduced the electronic complexity. The 501 uses one new high-speed A-to-D converter and a new high-speed D-to-A converter, both of which are time-shared between channels. The analog circuitry uses different reconstruction filters and higher quality audio components, while the video circuitry incorporates an "optimum video condition" control to optimize the unit for poor video recordings, as well as those made at slow tape speeds (Beta III or VHS SLP). The 501 is capable of playing back tapes that will cause an F1 or 701 to mute or glitch.

Sony also manufactures the **Nakamichi** DMP-100 processor, which essentially is a carbon copy of the F1. The primary differences include polypropylene audio input and output coupling capacitors that cost several times those of the F1's mylar capacitors, and sound noticably sweeter.

The **Sansui** X-1 Tricode processor has been in existence almost as long

DIGITAL RE-RECORDING AND REMIX OF VINTAGE HENDRIX TAPES USING 3M DMS AND JVC DAS-900

by Joe Gastwirt, JVC Cutting Center, Hollywood

According to Alan Douglas, founder of Are You Experienced Productions, and Hendrix' last producer, our goal in the reconstruction of *Jimi Hendrix Live At The Monterey Pop Festival* soundtrack was to produce a master tape to be used for a new film soundtrack, plus video, LP and Compact Disc release. Douglas, who has been working to keep Hendrix' music alive since his death, stated that his philosophy on this project was to recreate the live, raw, imperfect original sound of this concert. There were to be no overdubbed instruments or drum machines; we were simply to work with what we had.

We decided to mix down to a digital two-track tape machine, since it was agreed that if any advantages existed to analog they had already been embossed on the original multitrack tapes. We also did not want to compound any noise problems with additional analog tape hiss. Our overall goal was to use 1985 digital technology to restore and preserve the 1967 live sound. The two-track digital format we chose was a JVC DAS-900 processor, rented from CMS Digital, Altadena, CA.

Upon inspecting the original 1967 eight-track tapes, I knew we ran the risk of the tape falling apart after one or two good passes. Parts of the tape were already shredding, and the oxide was missing in a number of spots. In order to eliminate any further deterioration, I decided to transfer the original multitrack tape to one of Frank Dickinson's modified 3M DMS 32-track digital recorders. I chose this machine because of its superior sonic qualities, and Dickinson agreed to help with the different sync situations necessary to effect the audio transfers.

Mark Linett, who had already worked on the overall preparation of the *Kiss The Sky* album, and had mixed two of the songs, was asked to mix the *Monterey Pop* project at Sunset Sound's Studio, Hollywood. Sunset has a more than adequate monitoring system, as well as a custom control board equipped with Neve NECAM console automation, which quickly interfaced with the 3M digital multitrack. The studio was conveniently stocked with plenty of outboard gear, including GML parametric equalizers, several types of digital delay lines, and a live echo chamber. The relaxed feeling and the extremely helpful staff made for a good work atmosphere.

Analog-to-Digital Transfer

After spending about four hours hooking up resolvers, timecode generators, aligning the digital 32-track, and adjusting the eight-track head assembly to match the alignment of the

HENDRIX DIGITAL RE-RECORDING — *contined...*

original multitrack tape, we were ready to start the transfer. Because many extra tracks were available on the DMS, we bounced each of the original eight analog tracks to two digital tracks. The breakdown was as follows: two tracks each of audience, guitar, Jimi's vocal, bassist Noel Redding's background vocal, bass, drum, overhead drum, and 60-Hz sync tone, for a total of 16 digital tracks, plus one for SMPTE timecode (to provide for future synchronization with video, as well as for interfacing the 3M digital multitrack to the NECAM automation).

The 60-Hz sync tone from the original recording was put through a resolver, and used to ensure accurate speed matching during subsequent playback. This same 60-Hz sync tone was also recorded on the five-camera live film shoot from the concert, and is necessary for synchronizing the audio tracks to film.

Since the 3M DMS only holds 30 minutes of tape, and we were working with 45 minutes of original taped music, we had to divide the transfer into two parts. Convenient edit point with plenty of overlap time had to be found, in order to reconstruct the transfer into one continuous performance with consistent timecode throughout.

During the transfer stage, I was not surprised to find that, in spite of distortion, the vocals fading in and out, a drum mike disappearing and then breaking up while the sounds of the audience faded up and down, the tapes represented a truly remarkable performance, and one that also was a piece of history!

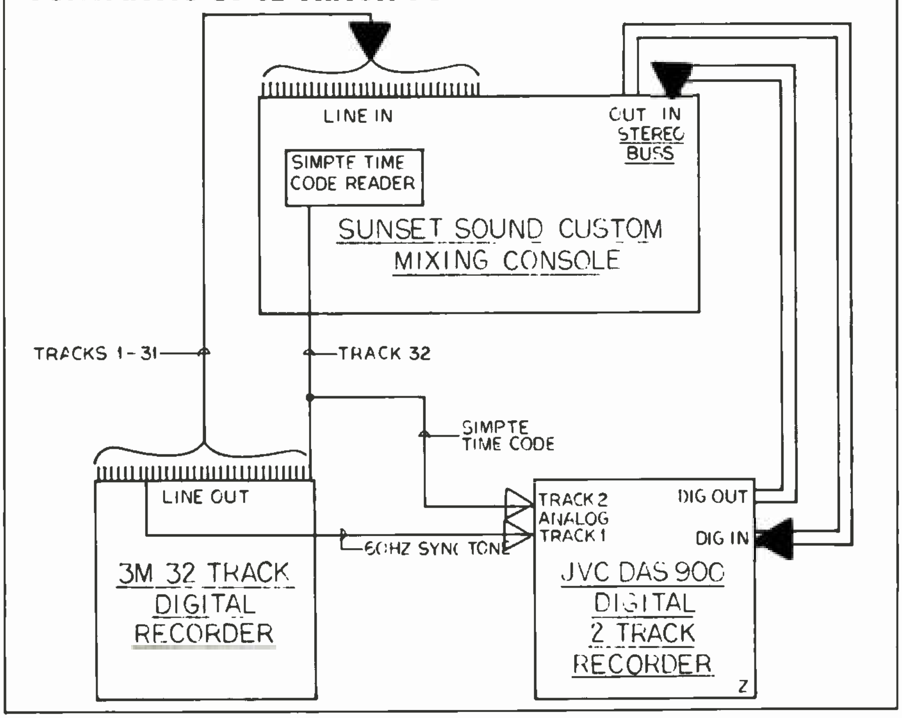
Remix Sessions

After taking several notes during this first listening, we were ready to set up the JVC DAS-900 two-track digital processor for mixing. The DAS-900 is a video-based system that works off either a preset internal sync, or a composite-video sync (sampling rate 44.056 kHz). Composite sync was taken from the 3M multitrack and plugged into the composite-sync input of the digital processor. At the same time, we routed the 60-Hz sync tone and SMPTE timecode to analog tracks one and two of the video recorder being used to record the digital information.

It was now time to begin the mix of the concert intro and the first song, "Killing Floor." We spent a lot of time on each track, trying different types of EQ, limiting, compression, digital delay and echo. After many painful hours, we finally came up with a mix that we all agreed upon. Linett and I took home cassette copies to listen to the mixes under familiar conditions. After listening again the next morning, we both decided that the first mix was a good try, but sounded a little "overdone." This opinion also made us aware that the first song was not a good place to start the mixing, since the sound was still changing on stage, and the flow of the concert had not yet been established.

We moved on to the second song, "Foxy Lady," to try a different technique. First we attempted to figure out the layout of the stage which, of course, was a very important consideration in terms of the placement of instruments relating to the film, and in keeping the audio as visual as possible. On this mix we used minimal EQ to keep the sound natural; the signal processing, digital delay and echo were used more to enhance the original sound

SCHEMATIC OF 32-TRACK TO TWO-TRACK DIGITAL REMIX



as the Sony units. A dedicated 14-bit processor, the X-1 is the first device to deal with the slowest VHS and Beta videotape speeds. The Tricode circuitry utilized in the X-1 effortlessly handles the poor video replay quality of slow-speed formats, and is the basis of the optimum video condition circuitry found in the PCM-501. To date, I have seen only two of these units, and am surprised that a more agressive marketing stance has not been taken by Sansui.

JVC has always shown a great deal of response to market input, and its VP-100 and VP-101 processors reflect the growing trends discussed later in this article, as well as potential applications beyond the original intended scope of the EIAJ-Format. A more rugged and "professional" model, the VP-101 is also a dedicated 14-bit processor, but has not been designed for used at slow VCR speeds — nor is the unit designed to be used with less expensive VCRs. JVC also manufactures a professional digital mastering system, the DAS-900. With the addition of a JVC interface, it is possible to digitally transfer from the EIAJ-Format VP-101 to the VP-900 16-bit processor, a feature currently offered by no other Japanese manufacturer of professional digital mastering systems. The VP-101 has switchable pre-emphasis and a sync output jack to control the speed of the VCR during playback. However, this jack does *not* accept external sync for video shoots, and limits the selection of VCRs to higher quality units capable of accepting external sync.

For those of us wishing to enter the fray at minimum cost, the recently introduced **Technics** SV-100 EIAJ-Format processor presents us with a 14-bit unit similar to the Sony F1, but without the 16-bit resolution. The unit has microphone pre-amps and can be used with a battery pack. Although the SV-100 is small and lightweight, a recent product review faulted the device for its lack of robust construction. Even though I have yet to see one, the combination of features and price makes this a most attractive unit.

Editing EIAJ-Format Material

One of the great bones of contention with respect to the potential use of EIAJ-Format processors in film-sound, audio-for-video and Compact Disc production, is the fact that all of them require that recordings be edited and transferred to professional digital formats, such as the Sony PCM-1610 or JVC VP-900. At present, two commercially available products exist to facilitate the video editing of EIAJ-Format digital audio without the need to convert it to a more expensive format for editing.

As many *R-e/p* readers may already be aware, the electrical and mechanical limitations of consumer VCRs necessitated the development of a more robust error-correction format to deal with increased random and burst errors. The encoded format spreads blocks of data over more than one video field with the result that, when fields of video are edited, dissimilar data blocks are joined to each other, causing data discontinuities and resultant glitches.

Electric Valve Communications, New York, manufactures an Editing Co-Processor (ECOP) that senses the data discontinuity, and removes it by digitally crossfading the incoming and outgoing material for a smooth audio transition. The device is a single circuit board that can be added to any EIAJ-Format processor via dealer modification. By correcting data glitches in the digital domain, ECOP enables edited tapes to be copied using the processor's digital copy feature; duped tapes will be free of audible glitches and data discontinuities. The device is "transparent" to any video editor, which might range in complexity from a simple control-

HENDRIX DIGITAL RE-RECORDING — *contined . . .*

of the tape, rather than to hide, mask or alter the sound.

Next we used the NECAM automation to mute any electrical clicks and pops, stage noises, intermittent buzzes and any other disturbing noises inherent in the live recording.

Our next task was to recreate the audience track between songs. Unfortunately, the original audience track had been faded up and down at the wrong times, and the applause cut off too quickly. We searched through other concert tapes to find applause that could be used to blend with the original applause track. We ended up with eight to 10 tracks of extra applause, mixed these together and melded them with the song. On each mix we allowed the applause to fade out and continue to the first few notes of the next tune, thereby providing plenty of overlap time to ensure smooth, easy transitions for the editing process. We completed "Like A Rolling Stone," "Rock Me Baby" and, my favorite, "Hey Joe," using the same method.

On the third morning we mixed "Can You See Me," "The Wind Cries Mary," and "Purple Haze"; in the afternoon we went on to "Wild Thing," and the burning-guitar segment. As many of you may remember, there was a massive amount of distortion on this last tune, which Linett and I were determined to eliminate. After some extensive detective work, we discovered that most of the distortion was caused by a tube drum mike that had crapped out in the middle of the show, but which nobody noticed. (Or, if they did, they did nothing to repair or change it!) After muting the bad track, we were left with only one usable drum track. The next few hours were spent trying such tricks as sending the sound back into a speaker in the studio, miking the speaker, and then mixing this with the existing drum to create additional ambiance.

After we felt secure that the drums sounded as good as possible, we fixed a few more noise problems and finished "Wild Thing." We then went back to the opening tune, "Killing Floor" and, following the same format and energy level as "Foxy Lady," ended up with a sparkling-hot mix. The session was complete.

Digital Editing

After a good night's sleep, I spent the rest of the day listening to the digital tape of each song, realizing I would not be able to rest until I heard the complete performance edited together as one, 45-minute show. Arriving at the CMS digital editing suite in Altadena at 11:00 that night, the first step was to hook up the DAS-900 digital editor so that it would retain the identical SMPTE timecode used in the original session. CMS matched the JVC

HENDRIX DIGITAL RE-RECORDING — *contined . . .*

timecode to the original timecode, by using the JVC TC-900 interlocking unit. Next, I had to be sure that the original 60-Hz sync tone was being transferred correctly to the new U-Matic master tape that would be the final production master. The actual editing process went very smoothly, because of the precautions and care taken during mixdown. The level of the introduction needed to be raised, in order to match the level of the first tune; the fader provided on the DAS-900 editing system made this a simple task to rehearse and execute. Most of the other edits used to join the show together were made during silence, crowd noise or Hendrix speaking.

Unlike most other music editing sessions, I had to concern myself with keeping the timecode consistent, rather than concentrating solely upon the music. For example, if the applause from "Killing Floor" ended at exactly 06:07;29, I had to find that exact timecode on the applause before "Foxy Lady" started, and match the two together perfectly. After spending some time level matching between applause tracks, making sure that each tune came in loud enough, and checking to see that all sync information was transferred correctly, I ended up with a tape that was ready to be mastered for LP, CD and to be synchronized with the film and video.

I do believe that there is an integral part of the human spirit that always strives for something good to be even better — not just for one's own satisfaction, but for everyone else as well. It was this desire that enabled Douglas, Linett and myself to find the energy and resources to keep the sounds of the great "Jimi Hendrix Experience" alive and audibly up to date. □□□

DIGITAL PRODUCTION

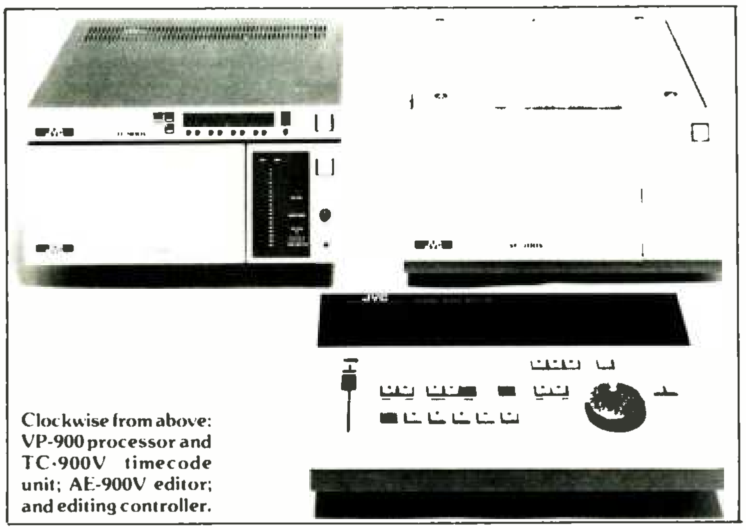
track model to a fully equipped CMX system, and has an editing accuracy of 16 or 33 milliseconds, depending on your choice of recording VCR.

HHB Hire and Sales, the London-based pro-audio supply and rental company, has developed a Computerized Logging Unit and Editor. Currently, CLUE is the only dedicated EIAJ-Format editing system, and is based on the use of half-inch Betamax VCRs controlled by a microcomputer running custom software. A full keyboard and VDU comprise the operator interface, with a PCM-701 replaying audio in real time. Edit points to a resolution of 33 milliseconds can be captured on the "fly," entered from the keyboard, or determined manually by bumping the tape forwards and backwards one frame at a time. Then the resultant edit point can be previewed before actually committing to it. While editing EIAJ-Format material, data glitches are removed in the analog domain through the insertion of a lowpass filter at the edit point to smooth over the anomaly. (For improved deglitching at the edit points, the Electric Valve ECOP is available as an option for CLUE.) Tape locations can be determined from the VCR's control track, or via an optional timecode reader board, and the edit decision lists stored to 5¼-inch floppy disk.

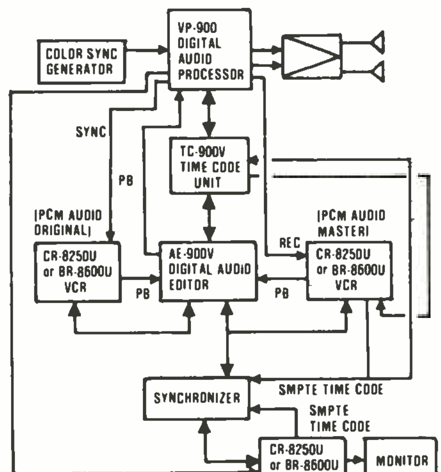
CLUE is particularly useful for arranging EIAJ-Format material that the user plans to transfer and then edit in 1610-format, for example. By working out the relevant edit points before the audio is transferred, you can greatly reduce the amount of (expensive) time spent editing in the latter format.

EIAJ-to-1610 Format Conversions

In the event that greater edit resolution is necessary, or for Compact Disc mastering, a conversion from EIAJ-to-1610 format must be executed in the digital domain. Utilizing the Sony DAE-1100 digital editing system,



Clockwise from above: VP-900 processor and TC-900V timecode unit; AE-900V editor; and editing controller.



VIDEO POST-PRODUCTION AUDIO MASTERING SYSTEM

The JVC Digital Audio Mastering System (DAMS) comprises two main components: the VP-900 16-bit, two-channel pulse-code modulation (PCM) digital audio processor, and the AE-900V digital audio editor. Because the processor incorporates a unique bi-parity (BP) recording format, and new error detection circuits, less expensive half-inch, VHS video recorders can be used to provide a two-hour recording capability, in addition to conventional ¾-inch U-Matic VCRs. Sampling frequency can be switched between 44.1 and 44.056 kHz. The AE-900V electronic audio editor has a quoted accuracy within 180 microseconds, and searching for an exact edit point can be achieved in one of three ways: manual cuing, automatic scanning, or direct address input. The unit confirms cut-in and -out points by recalling the signals stored in memory; a variable-gradient, crossfading function permits smooth continuity of program at the edit point.

Other components to the digital audio system include the RM-900 remote control, which operates both the audio processor and the companion VCRs; and the TC-900V timecode unit, which reads and generates SMPTE timecode and synchronizes it to JVC's own bi-parity timecode recorded digitally — a process essential for proper synchronization between the audio system and video equipment. The Digital Audio Mastering System is fully compatible with earlier JVC audio systems.

1610-format material can be edited down to the data-block level (1/7th of a video field, or 4.5 milliseconds), and is the accepted format for Compact Disc master tapes received by the majority of CD manufacturing plants.

Currently, three manufacturers are supplying devices for EIAJ-to-1610 conversion, a market that was created originally by **RTW**, whose products are distributed in the U.S. by **Auditronics**, of Memphis, Tennessee. RTW began format conversions in 1983 with the introduction of its original Studio Processor Set. As well as providing format transfer from F1 to 1610, the unit also features line-level analog interfacing, calibrated input attenuators, control of pre-emphasis and copy prohibition, and an expanded display showing the operation of the error-correction and concealment circuitry.

Recently, the PCM Set 2 and Set 3 have appeared, replacing the original standard bearer with new improved models offering expanded capability. PCM Set 2 allows format conversion from F1 to 1610, including all the aforementioned options featured on the original Studio Processor Set, plus switchable NTSC/PAL video standard, and RTW peak program meters. PCM Set 3 adds the capability for bidirectional format conversions: F1-to-1610 and 1610-to-F1.

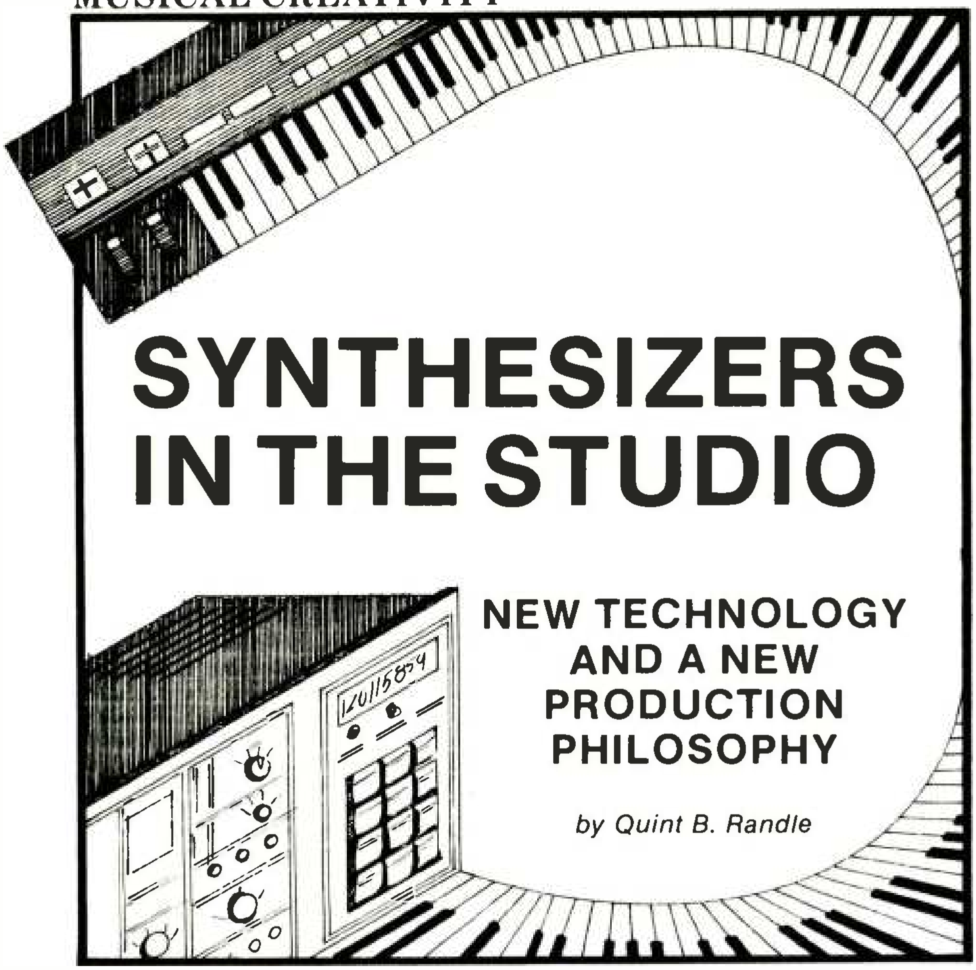
Audio+Design/Calrec, of Bremerton, Washington, offers a very clever modification package for PCM-701 processors that can be purchased as add-ons, or contained within a modified processor. Starting with the basic 701, an internal electronics package is added, along with new metalwork for the front and back panels that contain precision input attenuators, analog line-level interfacing, a choice of PAL or NTSC video standards, pre-emphasis and copy-prohibition control, full bidirectional format conversion, expanded error-correction indication, and (something lacking in the RTW units) Coincident Time Correction (CTC). When transferring material from F1- to 1610-format, the fact that a single D-to-A converter is time-shared (switched back and forth) between both left and right channels requires that an 11.34-microsecond delay be built into the F1. This delay should be corrected during digital-to-digital transfers to preserve monaural compatibility. (Since it utilizes separate converters for each channel, the PCM-1610 has no built-in time delay.)

Future developments by A+D/C include ADD-MIX, a digital fader and mixing system that allows the user to digitally mix two F1 sources, and also vary their levels during mastering, with a remote-controlled fader unit installed at the mastering console.

The most recent, and impressive, accessory for converting EIAJ-Format material to 1610- and the AES/EBU-format is the BW-102 Professional Digital Audio Interface from **Harmonia Mundi**, of West Germany, and currently distributed in the U.S. by **Audiotechniques**, New York, and **Audio Intervisual Design**, Los Angeles. Developed by Daniel Weiss, the gentleman that designed the hardware contained within the Studer SFC Sampling Frequency Converter, the BW-102 interface covers every possible combination of format transfers conceivable, in addition to a host of corrective measures. Based on a Eurocard mainframe construction, the unit allows the user to select input and output formats by installing appropriate modules in the cardframe. Modules are available for EIAJ-, 1610-, and AES/EBU-format input and output signals, as well as corrective modules that connect between the input and output modules to provide the capability of changing levels with a fader, interchannel phase reverse, right/left reverse, digital removal of pre-emphasis, or 11.34-microsecond interchannel delay. Also, and possibly the most useful feature of all, a digital highpass filter module is available for eliminating the distortion and poor edits that result from the presence of excessive DC levels in A-to-D converters during recording, or generally improper recording techniques. Future modules include a digital equalizer, a digital limiter/compressor, and a limited sampling frequency converter (44.1 kHz to 48 kHz, and back).

My personal interest in and experience with EIAJ-Format digital audio is based on two factors, neither of which have been addressed by the Japanese manufacturers: cost and portability. While EIAJ-Format processors are *not* built to professional standards, a factor that accounts for their low cost, with a little extra care and precaution they have turned out to be as reliable — if not more so — than professional processors, especially in their ability to play back tapes that are out of format specification. No portable professional processor exists and, at the present rate of development, we are unlikely to see one for several years; several EIAJ-Format units *are* fully portable, however, and the only way to record digitally in the field.

All of which argues strongly for the continued existence of the EIAJ Format, at least until some form of small, reasonably priced professional processor becomes available. □□□



SYNTHESIZERS IN THE STUDIO

NEW TECHNOLOGY AND A NEW PRODUCTION PHILOSOPHY

by Quint B. Randle

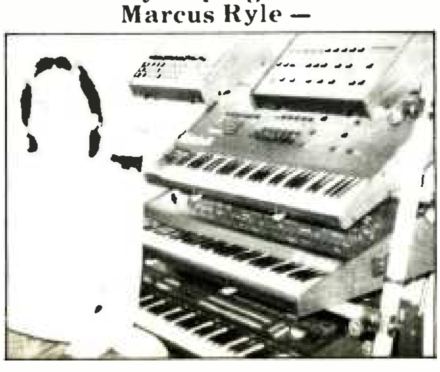
In the last year or two, a number of advancements have drastically changed the way in which engineers, producers, and players perceive synthesizers and keyboard instruments in general. It was just a short time ago that session players and studios alike were waiting for back-ordered Yamaha DX7s; and we now sometimes wonder how we ever got along without them. Another example of just how quickly this new synthesizer technology has been creeping upon the studio world is Chicago's *17* album. Most might consider that this 1985 Grammy-award-winner utilized MIDI to the extreme. But, while discussing the project with session engineer Humberto Gatica, we discovered that producer David Foster recorded several keyboard-bass parts in two or more passes. "At the time we started [May 1983] we could not MIDI the DX7 and Minimoog; we didn't have the software to do it," the engineer confides.

Realizing that the dawning of a new era in keyboard technology has been somewhat overwhelming to the recording industry, it may be useful to step back, take a deep breath, and approach the subject anew — erase the blackboard, in a sense, and pull out a new piece of chalk. In this two-part series, with the help of engineers, technicians and session players, I will approach the subject of keyboards-synthesizers, and hopefully clear up some of the new, as well as old, problems and myths that have developed. Part one will re-define some of the old terminology, and define and examine some of the new variables that affect the engineer/producer/player in a multikeyboard situation. The second installment will delve into some of the deeper, more complex hands-on problems one faces when as many as eight keyboards are connected via MIDI, or synchronized to tape — hopefully providing some solutions.

Types of Keyboard Synthesizers

Ask any session player or synthesizer programmer what it takes, equipment-wise, to survive in today's market, and he or she will probably say that you need at least one keyboard from the following three categories: analog-based (or subtractive synthesis); digital FM-based (or additive synthesis); and sample-based (digital PCM sampling).

— Player/programmer Marcus Ryle —



The ***Analog*** synthesizer has been around the longest, but is well worth re-defining for our purposes. **Marcus Ryle**, a session player/programmer and former technician at Oberheim, describes basic analog synthesis as "circuitry that oscillates at a particular pitch. A voltage-controlled oscillator [VCO] is usually employed to allow varying voltages to change the pitch of the produced sound. Before reaching our ears, the signal passes through a [voltage-controlled] analog filter, and an analog amplifier."

Generally, this mode of operation is referred to as subtractive synthesis: a sawtooth wave passes through an analog filter, and whatever the programmer doesn't want included in the finished audio is subtracted from the initial waveform to produce the required timbre.

The rest of the circuitry in an analog keyboard, Ryle continues, depends at what point in the history of analog synthesis the unit was developed. "Basically, all analog synthesizers on the market nowadays are controlled digitally. For example, in the case of the Oberheim Xpander or Matrix 12, the audio section is all that's analog; everything else is digital — all the factors which modify the original analog-produced wave are controlled via a microprocessor."

The fact that these variable commands are encoded digitally means they can be memorized and stored by the synthesizer's microprocessor.

Also, depending on how sophisticated the digital controls are on an analog synthesizer, it could mean that all of the circuitry can be automatically calibrated by the built-in microprocessor; you don't have to worry about the keyboard going out of tune in the middle of a take, for example, or waiting for it to tune up. In addition to this, and again depending upon sophistication, Ryle continues, "Digital technology allows a great deal of flexibility in terms of modulation that might not have been possible previously. Modulation can be created through software control, rather than hardware itself.

"On the Xpander and the Matrix 12, there are five envelope generators and five LFOs [low-frequency oscillators] per voice that don't really exist in the hardware; they've been created in the software by the computor."

Ryle also points out that there are a number of synthesizers employing digital oscillators, which create the

The inventors of the digital drum machine now offer you the most sophisticated compositional tool ever created. The Linn 9000.

PPG WAVE 2.2▲ ▲SEQUENTIAL PROHET 600

NED SYNCLAVIER II▲ ▲E-MU SYSTEMS EMULATOR II

initial audio signal, but the signal is then sent through analog filters and analog amplifiers. For instance, although the Roland JX-3P and the JX-8P utilize such digital oscillators, the finished audio is created by passing the initial wave through subtractive analog filters. The main difference is that the device creating the initial waveform digital as opposed to analog; subtractive synthesis is still the *modus operandi.*

"Whether a subtractive synthesizer uses analog or digital sound-producing capabilities is really just going to give you different types of sounds," the consultant concludes.

Digital FM: Stated simply, a purely digital synthesizer generates a "sound" with numbers, so to speak, by adding different sinewave values together to obtain desired sonic characteristics. Towards the end of the creation process, these numbers are fed into a digital-to-analog converter (DAC) so that, when amplified, our audio sensors can make some sense out of it.

Yamaha's series of digital FM synthesizers, including the DX1, DX7 and DX9, probably represents the most widespread implementation additive synthesis in use today. **Kevin Bierl,** a Yamaha technician, explains how the DX7 works:

"The digital sound source, or operator, is itself a sinewave," he says, "but there are six of them on the DX7 (four on the DX9). At this point, depending on what kind of sound the programmer is trying to attain, the six operators are digitally programmed to affect one another via the unit's 32 algorithms [software-controlled processing steps] — in essence, how they will be added together.

"Depending on the algorithm," Bierl continues, "that tells the computer which of these six operators are 'Modulators,' and which are 'Carriers'; a Carrier is what you hear, a Modulator is what you don't hear — you can only hear the Carrier, and the imprint the Modulator(s) add to the Carrier."

Using such algorithms, the programmer sets the parameters, and thereby develops waveform characteristics: different ratios between Modulators and Carriers; frequency of modulation; how powerful an operator will be; envelope variables; in what order the operators will be used; etc.

"For example, we use a ratio of two to one — the Carrier is twice the frequency of the Modulator — to come up with what is a basic square waveform, which is good for a reed-like sound."

But all the while, Bierl explains, "it's still numbers to the computer. Finally, at the very end of the process, the numbers are fed into a DAC and the new analog sound is amplified."

Digital Sampling: The basic technical distinction between synthesis and sampling is that the former technique creates a sound from scratch, as it were, using the previously described methods. Sampling, on the other hand, uses a computer to digitize and store a sound introduced to it from an external source; and then, in its simplest form, plays the recording back at various pitches.

"As far as the development of sampling is concerned," says **Marco Alpert** of E-mu Systems, "it really began to become known about five or six years ago, with the first Fairlight CMI, more than anything else. And then we [E-mu Systems] like to think that the original Emulator I was the first instrument designed to provide [cost-effective] user sampling."

On the market today are a range of sampling synthesizers, including the Fairlight CMI, Emulator II, New England Digital Synclavier, Kurzweil 250, PPG Wave, and Ensoniq Mirage. Although each of the manufacturers varies somewhat in its method, Alpert provides a flow-chart explanation of user sampling:

▲MIRAGE ENSONIQ

"The sound enters the device in analog form from a microphone or line-lever source, or directly from another instrument. It is then fed through an analog-to-digital converter (ADC), which measures the incoming voltage a prescribed number of times a second; for example, the Emulator II samples 27,500 times a second.

"Each time the ADC measures the voltage, it checks to see at what level the voltage is, gives that level a number value, and stores it away in memory. It then repeats the process, stores the level numerically, and so on. It does that for as long as the sound lasts, or until the available memory is depleted."

At this point, the keyboard/computer has a string of 12-, 14- or 16-bit numbers representing levels of the initial incoming voltage taken at 36-microseconds intervals (in the case of the Emulator II), that can also be stored on cassette tape or floppy disk.

To play back the original sample, the synthesizer's microprocessor transfers a 12-, 14-, 16-bit number into a DAC which, after filtering, returns

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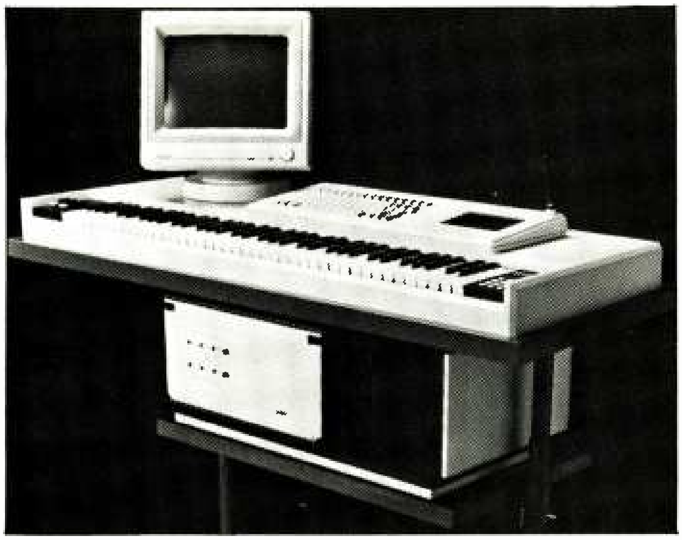

FAIRLIGHT CMI SERIES III▲ ▲KURZWEIL 250

SYNTHESIZERS IN THE STUDIO

it to an original voltage wave.

When it comes to reproducing a sound at a pitch other than the original, there are several methods available. The most common according to Alpert, is that of speeding up or slowing down the clock rate. "If you want the pitch to be higher, instead of reading the stored numbers 27,500 times a second during playback, the computer reads at a rate of, perhaps, 30,000 per second. Or, if an octave higher is desired, the reading rate is then doubled to 55,000 times a second. The faster the numbers are spit out of memory, the higher the pitch; a slower rate of reading the memory produces a lower pitch."

Technicalities aside, Alpert says that an important consideration with any sampling keyboard is its sampling rate — how many times a second the device measures incoming voltage.

"Sampling rate defines the frequency response of the system; the highest frequency that can be reproduced accurately is one half the sampling-rate frequency. In other words, if a system has a 30 kHz sampling rate, then the highest frequency you can reproduce is 15 kHz."

While some sampling keyboards allow the user to increase the sampling time (length) by reducing the sampling rate, this causes a reduction in frequency response, he concludes.

MIDI Takes Control

While a continual perfection of these three synthesizer types during the past decade has had a tremendous effect on the new keyboard philosophy, nothing has changed the way we view synthesizers more than Musical Instrument Digital Interface. With MIDI, a synthesizer programmer now has an almost unlimited number of variables available while creating a particular sound. Instead of dealing with only one synthesizer at a time, the player can now mix and match different keyboards, using the strengths of each to invent his or her own unique sound. And, as stated in its proper name, MIDI is not just intended for keyboards only, but for virtually all electronic musical instruments. Non-keyboard devices, such as sequencers, drum machines and, most recently, reverb units, can now be interfaced with a keyboard(s) in real time to create a multidimensional, multifunctional musical incarnation.

The unique audio combination of several synthesizers is, of course, the most common end result to all this, but MIDI is the all-important means to that end. Part two of this article, to be published in a subsequent issue of *R-e/p*, will examine a few of MIDI's operational imperfections, but for now a brief description of this communications standard and its capabilities as related to synthesizers (or those purported to be offered by manufacturers) is in order.

Bob Moore, president of Hybrid Arts, a company that produces, among other musical software, a MIDI sequencer for use on personal computers, draws this analogy when definning MIDI: "In computers there are interface standards that allow systems to communicate [with themselves and external devices] virtually anywhere in the world, the most common of these standards being RS232. MIDI is basically the same thing, but for musical instruments; it allows electronic music synthesizers to communicate with one another.

"It's essentially just four wires: two in each direction — a bi-directional, serial communication of MIDI In and MIDI Out information. In other words, a keyboard can receive information and, at the same time, it can put out [eight-bit digital] information [at a transfer rate of 31.25 kbits per second]."

— Hybrid Arts' Bob Moore —

MIDI can transmit all kinds of data between synthesizers and other MIDI-equipped devices, including commands for selecting a new voice, pitch change, modulation change, breath control, sustain control, after-touch, clocking information, various sequencer commands, plus a variety of system-exclusive commands that are up to the manufacturer to define.

By far the most common command sent from a master keyboard to slaved units is *Note On* and *Note Off*. "Because of the MIDI standard," Moore offers, "even though the oscillators are different, and the way another keyboard might generate its sound, Note On and Note Off are standard communication commands. For the most part, Note On and Off are three bytes of information each: a Note-On-and-Note-Off event — that is, a single stroke of the keyboard — is a six-byte event. "Note On is three bytes long; the first byte communicates the [MIDI] channel code; the second byte is the key value of the note; and the third byte is its velocity." [For a complete description of the various hexadecimal codes for MIDI commands, refer to page 125 of the December 1983 issue of *R-e/p-Editor.*]

. . . continued overleaf —

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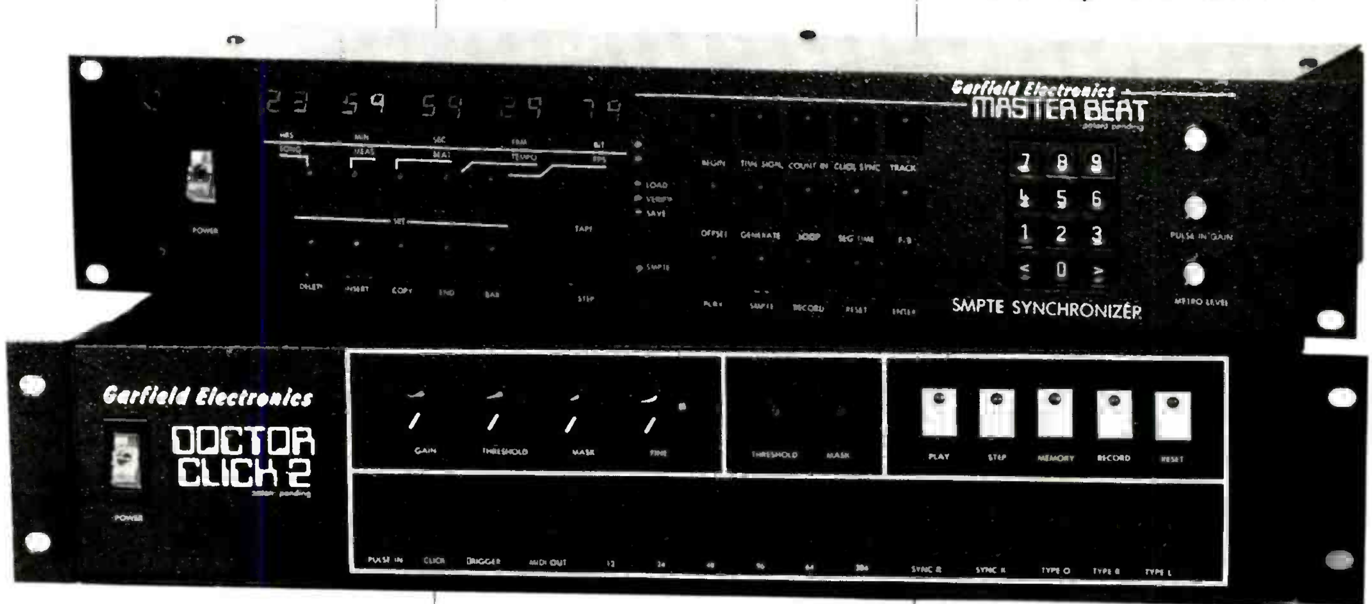

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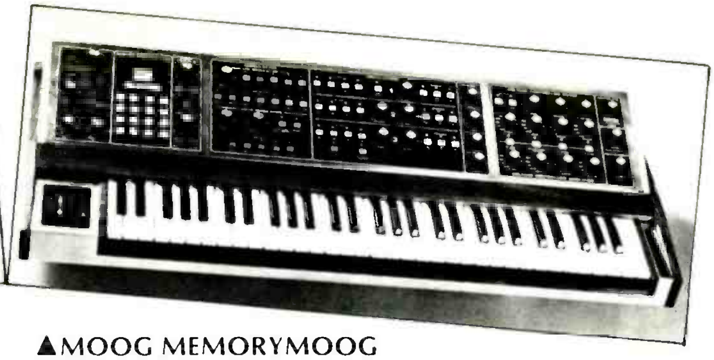


ROLAND JUPITER-6 ▲ ▲MOOG MEMORYMOOG

SYNTHESIZERS IN THE STUDIO

The MIDI standard provides 16 independent channels over which to accomplish the transmission of data. Each MIDI synthesizer only recognizes which MIDI channel (1 thru 16) it has been assigned to recognize, or to receive on.

"For instance," Moore explains, "when a sequencing computer puts out 16 channels of MIDI information, a keyboard will only pay attention to whatever information is communicated over the particular channel it is assigned to; it won't do anything until it sees something within the byte stream it is 'told' to recognize."

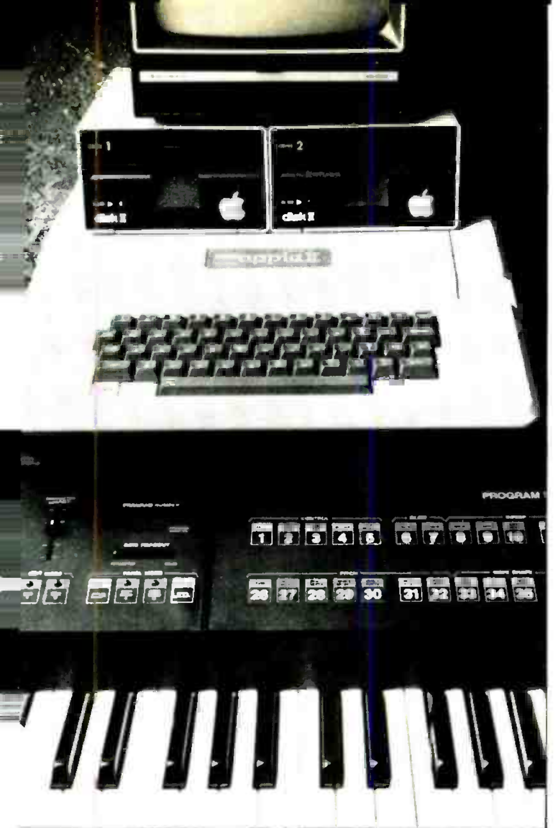
▲YAMAHA DX7 FM/DIGITAL SYNTHESIZER

When sequencing from an external computer, the master keyboard on a certain channel is patched to the computer's MIDI In and MIDI Out. Then the computer, in effect, disperses that information to other keyboards on varying channels.

Some sequencing systems can be connected via MIDI Clock to, in a sense, sequence one another, thereby increasing the number of available tracks by multiples of 16 or 32. And Hybrid Arts, for example, now has a system on-line that allows the transmission of musical parts as MIDI information via telephone modem. "We're also trying to help Motown put together a network of computers so that their publishing group can communicate throughout the city with all of their musicians and composers," Moore explains.

Sync Codes

Another important advancement, and one that has more recently had a direct effect on keyboards, is the ability to synchronize different devices in real-time, or to a timecode track on tape. Some synthesizers provide on-board sequencers, while others allow sequencing and triggering externally via MIDI or CV trigger inputs. The use of sync codes is generally referred to as "clocking": a master clock on one device is used to drive the clock(s) on another device(s) to communicate tempo, start points, etc. In approaching the more complicated MIDI and



RHODES CHROMA ▲

sync problems to be described in part two of this article, it is important to understand the various sync codes in use today, and why some are more reliable than others.

"This whole business of syncing started with the film industry," says **Jim Mothersbaugh**, technical administrator at Roland. "While recording a film soundtrack, a conductor needed some way to keep track of, and remain in time with, the film. At first, a simple metronomic beat was used; and, after some time, a visual cue for the conductor was developed called a 'streamer.'

"Today, after three decades of electronic improvements, the music industry has adopted, in one way or another, most of the timing/sync codes that were developed along the way by the film and TV industries."

Pulses-Per-Quarter-Note. The most common forms of clocks found on keyboard sequencers and related devices are 24, 48, and 96 PPQN. Mothersbaugh explains that on-again/off-again voltage-spike pulses are sent every quarter-note [or beat] from the master clock to the slave. This, in turn, allows the devices to be synchronized together, the slave following the master.

Logically, a 48 PPQN clock is sending or receiving 48 pulses, rather than 24, each quarter note, and so on. While the higher pulse rates provide a slightly tighter resolution, Mothersbaugh claims that there is little difference in overall reliability; the reason various manufacturers (mainly Roland, Oberheim, and Linn) used different clock rates was to create system exclusivity.

Frequency Shift Key. After the development of PPQN clocking, a technology called Frequency Shifted Key (FSK) was devised specifically for tape synchronization.

"FSK is very similar to PPQN clock pulses," offers Mothersbaugh," in that there's a rapid succession of pulses. But, in this case, rather than using on-again/off-again voltage spikes, two frequencies are being modulated against one another at 1.2 and 2.1 kHz. The result is a smoother, less fluctuating signal that can be recorded at a lower volume on audio tape, reducing the chance of crosstalk to adjacent tracks on the machine."

Although not as widespread on electronic instruments as PPQN clocks, FSK is still used on some devices today.

MIDI Clocking. This form of clocking is similar to the PPQN method — on/off pulses — but operates using MIDI specifications and MIDI-controlled hardware. The main drawback with MIDI clocking, according to Mothersbaugh, is that "MIDI operates at a frequency of 31.5 kHz, which is obviously too high for a tape machine to handle." As a result, MIDI clock commands need to be effected in real time.

▲ PASSPORT SOUNDCHASER

The problem with [PPQN, FSK, and MIDI] clocks at this level, is that they are a "dumb" tempo indicator; they can do nothing more that specify tempo. For example, they can't tell devices when or where to start up and stop again; they can't tell what beat and what measure they are at, etc. The key is the ability to go one step further, and be able to sync and punch in the middle of a song. With the dumb-clock methods, you have no reliable way — other than starting at the beginning of a song [and have the

— continued from page 55 . . .

SYNTHESIZERS IN THE STUDIO

slaved device count pulses from the downbeat] — to have everything link up.

SMPTE Code: The first "smart" code was devised by the Society of Motion Picture and Television Engineers in the Sixties as a standard of synchronization for the film and video industry; timecode is now a standard within the recording industry as well. In essence, SMPTE timecode is just a linear number sequence opposed to an on/off code. When laying SMPTE code to tape, the slaved device is counting out, in a very fast and fine revolution, minutes, seconds, frames, and bits (1/80 of a frame). With the advent of SMPTE timecode synchronization, start, stop, punch-in, punch-out points, etc., can be referenced anywhere in a tune, not just relative to the beginning.

The problem is that most electronic musical devices (virtually all conventional synthesizers) do not provide on-board SMPTE timecode-reading capabilities. To alleviate this problem, a number of companies are manufacturing dumb clock/SMPTE interface units that translate SMPTE timecode into PPQN codes, etc.

"In a musical setting," says Mothersbaugh, "it takes an interface box like the Roland SBX-80, Friend Chip SRC and Garfield Electronics Master Beat, to do this advanced type of synchronization. These devices read timecode off tape, and mathematically convert that to clock/tempo information instruments can read." (For a detailed explanation of the timecode to MIDI and sync clock process, see Bob Kinkel's article on the Roland SBX-80 and Friend Chip SRC in the April issue of *R-e/p*.)

MIDI Song Pointer: Another MIDI specification on the horizon, deserves a mention. The Song Pointer function provides many of the luxuries previously available only through SMPTE timecode. "With Song Pointer, we can send not only clock and tempo information through MIDI, but it can say, 'Okay, you sequencers and drum machines, you're starting up at measure 32, beat two, at 120 beats-per-minute'," Mothersbaugh reveals.

The Audio Signals

The net result of these new technologies is generally the engineer dealing with multiple audio signals from a "Keyboard System," rather than from one synthesizer at a time.

"It almost changes your philosophy on what you consider a synthesizer to be," says player/programmer **Paul Fox**, whose credits include projects

— Player/programmer Paul Fox —

with the Pointer Sisters, Commodores, Natalie Cole, Thelma Houston and Cock Robin. "Before, you were dealing with each individual axe as its own sound, and combining the sounds [on tape]. Nowadays, I seldom put my system together without first figuring out the best way to create total sound."

A few years back, Fox says, he would bring along two, maybe three, synthesizers at the most. "You used to come in and put your Prophet up on the producer's desk, and that was that; you basically fed the engineer your audio signal, and that's basically where you lost control — unless you developed a rapport with the engineer."

That's ancient history now. For 90% of the time on sessions these days, Fox uses more than one synthesizer to create a sound before going to tape. "It's like 'MIDI-mania.' You have to be open to saying to yourself, 'Okay, is this *too* much?' With just about everything, I use component synthesizers. Recently, I did a horn sound with seven or eight different synthesizers: I had two horn samples; I had six DX7 patches [from the new TX-816] — some of them were duplicates, but were detuned a bit."

From a sonic perspective, MIDI-mania has presented two obvious problems: How many tracks are there available on the tape? And should the balance betweeen the different components of the finished sound be committed to tape? Some studios are dealing with these questions by providing in the control room sub-mixer especially for keyboards. This additional mixer provides a stereo feed to the console exclusively for synthesizers, and gives the engineer and producer more control, as well as confidence, in committing a particular balance to tape.

"A lot of guys, especially TV and film players," Fox adds, "are bringing along their own mixers to sessions. So what he or she is sending to the board is basically a left and a right output. And, depending on how much gear the guy has, this may include his own effects rack. A setup like this enables the synthesist to really go for the sound he or she hears in his or her head; they can treat the entire setup as one synthesizer — a 'studio within a studio'."

Another simple problem that can occur in a multiple-synthesizer setup is the noise factor. Some models put out a ridiculous amount of ground noise when surrounded by other units. Over and above any serious in-building ground problems, Fox says the best way to combat this is by using a ground lifter (three-prong-to-twoprong adaptors). "I would suggest every synthesist always carry five or ten with them, because you're always losing them."

In a subsequent issue of *R-e/p*, I'll move on to examine the idiosyncrasies and set-up problem encountered by several studios and session players in creating and maintaining their own unique multiple-component synthesizer system. ■■■

MIDI UPDATE

MIDI-Equipped Synthesizers, Sequencers and Software at the Summer NAMM Exhibition, New Orleans.

by Bobby Nathan

For me, one of the more striking MIDI innovations on show at the recent Summer NAMM Exhibition, held in early June at the New Orleans Convention Center, was the **Music Data's** Midi Delay — not the time delay inherent to most MIDI-equipped sequencers and synthesizers, but one that can be programmed! Midi Delay is a software package created by Lance Ono for the Apple IIe. A total of 16 programmable presets are featured for delay parameters, including feedback (which is really the number of repetitions); amplitude (the actual level in amplitude of each repetition in relation to the previous one measured in a plus and minus range); pitch change (the interval of pitch between each repetition); and the MIDI Channel that the repetitions will be triggering. Unlike analog and digital delays utilizing A/D converters, the bandwidth of the Midi Delay is the same as the MIDI device being triggered by the unit. The plus and minus amplitude programming makes for repetitions of the same amplitude, ascending amplitude, or the standard descending amplitude that we know so well from standard analog and digital delays.

Another innovative first was **Korg's** DW-8000 eight-voice velocity/pressure sensitive keyboard with built-in digital delay. The provision of a built-in delay unit might very well start a new standard in including outboard gear into synthesizers. The need to add delay, reverb and ambience to synthesized sounds is already a well-known practice in studio synthesis. The DW-8000 keyboard also features 64 presets, each having its own individual programmable delay settings. Parameters include delay amount, feedback, delay level, and an independent LFO section with rate depth and waveform. There are also two digitally-controlled oscillators (DCOs) that can produce 16 digital waveforms, including the standard sine, square and sawtooth waves. For live performance and/or studio sessions, the self-contained digital delay adds all the right dimensions to complement each of the contained presets. In the near future, Korg will also release the EX-8000, a rack-mounted version that encompasses all of the same features as the DW-8000, except for the keyboard.

Also in the keyboard corner, **Yamaha** added the DX-5 synthesizer to its FM Series of keyboards. The DX-5 is, in actuality, two DX-7s in one convenient package. The keyboard features the same action as found on the DX-7; two cartridge slots and the ability to program function controls has been added. Balanced line outputs for left, right and mix are available on both XLR and ¼-inch phone jacks. The DX-5 has basically all the features of Yamaha's super DX-1, but for a much more affordable price. The company also introduced the KX-88 Master MIDI Controller Keyboard, which features programmable splits with separate MIDI channels. There are also four programmable sliders that can be set to your favorite performance parameters of any of the DX and TX Series synthesizers. The KX-88 has a 88-note, wooden-key velocity keyboard, and could very well be perfect for the player who wants true piano touch and response. The synth is the perfect complement to playing and programming the TX-816 rack, which houses eight DX-7 modules in rackmount unit with programmable function controls.

A new entry into the MIDI scene was **Roland**'s GR77B MIDI Bass Guitar Controller and Synthesizer. The GR77B resembles the already standardized GR700 Guitar Synth in appearance, but has improved tracking. Unlike the GR700, the GR77B features a separate microprocessor for each string, making for the much improved tracking. Because of the lower frequency range of a bass guitar, the triggering should in reality be slower. The unit includes a synthesizer section based on a JX-8P with 64 presets, a cartridge slot and MIDI Out and In.

On the MIDI guitar scene, **Octave Plateau** unveiled its long awaited guitar controller, which features a fretboard wired for fast and accurate response. The guitar itself features a Voyetra-style keypad (telephone type) to call up program and parameter changes for whatever synthesizer is being controlled. The guitar comes with a phanthom-powered box that provides life for its active electronics and MIDI-Out jack. There are also three programmable levers that can be set to favorite performance parameters.

IVL Technology introduced its Pitchrider 7000, a MIDI guitar interface, which comprises a hex pickup that can be mounted on almost any guitar, and an electronic "brain" housing six individual Pitchriders. Each string has its own converter,making for quite accurate triggering.

In the electronic drum department, **Simmons** introduced the analog/digital SDS-9 drum set, a five-piece kit featuring digital samples for the snare, analog kick, and three analog toms. The snare channel actually has three different digital samples — one for the center of the head, and two for the rimshot — thus making for a dynamic, real sounding snare. Different tunings, release times and noise balances can be preprogrammed into 20 memories. The SDS-9 also featured a first in electronic drums: MIDI-In and -Out jacks that enable the device to trigger and be triggered thru MIDI, various sequencers, synthesizers, and other MIDI drum machines.

Speaking of drum machines, **E-Mu Systems** introduced the SP-12 sampling percussion, a 12-bit sampling velocity MIDI drum machine. The SP-12 is 99-segment/99-song machine with 24 internal sounds stored on EPROM chips, and which can also store eight sampled sounds in non-volatile memory. Two different models are planned for market release: the basic unit with 1.2 seconds of sampling time divided among your eight samples; or the Turbo version, which has its memory beefed up to five seconds of sampling time, and the segment/song memory expanded from 16 to 64 Kbytes. In many ways the SP-12 has incorporated many of the Emulator II's features into its front-control panel; there are eight velocity tap pads with up to 32 levels of velocity that can be assigned to any of the internal or sampled sounds. Any sound can also be assigned to all eight pads, as in the case of a tom sample. MIDI In and Out capability will allow the SP-12 to control and be controlled by many various MIDI-equipped devices. The unit stores sequences and sampled data via its cassette or optional Commodore disk-drive.

Having trouble learning FM technology to program your Yamaha DX-7? Well, **Jellinghous Music Systems** just might have an answer for you. JMS has taken every parameter of the DX-7 and represented it by either a switch or knob on a cleverly laid out analog-style panel. Even if you've already learned to master DX-7 editing, the JMS DX-7 programmer can probably spawn new and interesting patches.

In the MIDI-equipped outboard gear corner, **Yamaha** introduced the REV-7 digital reverb, which has stereo inputs and outputs, a three-band, non-programmable, quasi-parametric EQ section, 30 presets (including halls, rooms, flanging, chorusing, reverse effects, and a number of presets designed for specific musical instrument applications), and 60 user-defined variations of the 30 original presets. Program changes can be controlled via MIDI; there is also a hand-held remote that includes some of the more important programming controls. The REV-7 has basically all the same features of the REV-1 — I played with both units in the Yamaha booth at NAMM, and found that I could basically duplicate reverb patches from the REV-1 on the REV-7.

Europa Technology introduced the

Window Recorder, a 16-bit sampler with MIDI triggering capabilities, available in three-, six-, 12- and 24-second sampling time versions. After a sample has been recorded into the Window Recorder, it can then be truncated — that is the front and rear of the sample removed to correct for slow attack and/or unwanted parts of the sample. The WR also has the ability to play a sample backwards. Another innovative feature is the unit's overdub mode that enables infinite overdub on top of the original sample. (The only catch is that if you're not happy with the final balance you're stuck.) The overdub feature can be most useful in stacking up snare drums to produce a really fat snare sound. Samples can also be tuned from the pitch at which they were sampled; a loop function will play back the sample continuously — a feature useful with percussion riffs, etc.

J.L.Cooper introduced an interface for the Quantec Room Simulators that adds MIDI program-change capabilities, etc. The new interface is a most useful feature for live applications, and in the studio, with a MIDI sequencer, can add automated reverb capability to your mix.

In the software corner, the Apple Macintosh finally has received some of the software development it deserves. Of all the Mac programs at NAMM, Total Music by **Southworth Music Systems**, a 99-track, 16-MIDI channel per track MIDI recorder, was the most complete integrated package available. Total Music allows the individual editing of each MIDI channel, either with graphics (a graphic representation of a track's note durations, velocity, and etc.), or as notes on staves. Music can be recorded via a MIDI keyboard, or by placing notes on staves Macpaint-style. After notes have been written on the staves, they can be instantly played back on your MIDI keyboard. The recorder features many different quantization values, and can also be programmed to record in step mode. The software includes a score-printing driver routine that can accommodate Apple's Imagewriter or Epson printers with the Mac-Epson connection; many other printers can be interfaced as well. All the menus are self explanatory, making the need for a manual almost non existent. Southworth has also developed its own Macintosh MIDI interface, which includes two MIDI Ins. With the two MIDI Ins, a drum machine's MIDI clock can simultaneously control Total Music, while MIDI data is recorded from a MIDI-equipped keyboard.

Opcode introduced the Midi Mac series of software for the Macintosh. The company's sequencer software includes 26 sequences of 10 tracks each; features such as loop, transpose and track mute are included. Tracks can be edited similar to autolocating on a conventional multitrack, and songs are created by joining sequences together to form new sequences. A separate Patch Librarian program can store any MIDI system's Exclusive data onto one disk. Synthesizers supported by the Midi Mac software include the Yamaha DX-7, Oberheim OB-8, Casio CZ-101, CZ 1000, CZ-5000, plus the Roland Juno 106 and JX-8P. Opcode has also introduced a custom-designed MIDI interface that can be used for both the Midi Mac Sequencer and Patch Librarian.

Digidesign's Sound Designer software for the E-mu Systems Emulator II digital keyboard provides gives the EMU II with capabilities beyond many of the higher priced computer systems. Complex looping and truncation start and end points can be displayed, greatly simplifying editing on the Emulator II. The Sound Designer menus resemble Macpaint, and all editing is controlled via a mouse. Cut and Paste features allow the attack of a trumpet to be pasted onto the envelope of a piano, for example. Via various menus, every function of the Emulator II's front panel is displayed graphically and numerically, for a quick status of where the filter, ADSR, and all the other parameters are set. When editing a waveform, you can make good use of the zoom magnification functions for precise editing. Waveforms, filter curves, and envelopes can also be hand drawn via a mouse — a feature that can virtually eliminate glitches caused during looping.

■■■

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

Northeast:

□ **TROD NOSSEL RECORDING STUDIOS** (Wallingford, CT) has installed an **Aphex Compellor** compressor/limiter, a **Loft Model 410** compressor, **API 554** parametric equalizers, a **Pearl** electronic drum set, plus **Yamaha NS-10s** monitor speakers and a **DX-7** FM synthesizer. Available for rental is a **Yamaha QX-1** with TX-816 system. *P.O Box 57, Wallingford, CT 06492. (203) 269-4465.*

□ **BROCCOLI RABE RECORDING COMPLEX** (Fairfield, NJ) has installed a new **Time Line Lynx** timecode synchronization system to its 48-track facility. In addition, two staff engineers have been added to the studio's staff — **Bill Berends** and **Ed Dougerty**. *15 Gloria Avenue, Fairfield, NJ 07006. (201) 575-7460.*

□ **DREAMLAND RECORDING** (Kingston, NY) is a new 24-track complex built by **Acoustic Spaces** and **Bija Productions**. Constructed in a "historic" church, the 800-square-foot control room/studio incorporates lead shielding in all walls, an "acoustically coherent" isolation booth, and video and telecommunication capabilities via computer control, plus a 40- by 45-foot ambient room with a 34-foot high ceiling. The control room features an automated 40-input **API** console with patchbay, **Valley People 65k** mixdown computer (containing 36 Fadex modules), **Sony/MCI** 24-track and **Studer A800** multitracks linked to a **Cipher Digital BTX Shadow** synchronizing system. The following monitors are supplied within the facility's control room: **UREI**, **Ed Long TA-3**, **Yamaha NS-10**, **Auratone**, and **JBL** models. Outboard gear comprises **Lexicon 224** and **PCM-60** digital reverbs, **dbx** noise reduction, **Tektronix** and **Pultec** tube equipment, and **Studio Technologies Echo Plate III** reverb. Company spokesman **Phil Miller** says that the new complex offers audio/video duplication services, rehearsal space, indoor/outdoor lounges, and video production capabilities. *Road #3, Box 288, Route 28A, Kingston, NY 12401. (914) 338-7151.*

□ **COUNTERPOINT RECORDING STUDIOS** (New York City) has completed a major upgrade with the acquisition of two **Solid State Logic 6000E** consoles, each with **Total Recall** automation, a pair of **Otari MTR-90 Mk II** 24-tracks, four Otari MTR-12 two-tracks, and an assortment of signal processing gear from **AMS**, **Eventide**, **Lexicon**, and **Marshall Electronic**. Aside from the SSL boards, all equipment was supplied by **Martin Audio**. In addition, Studio B has been enlarged, and a variety of cosmetic and acoustical improvements have been built into both studios, says the facilty's owner **Jerry Ragovoy**. *723 7th Avenue, New York, NY 10019 (212) 398-9550.*

SYNTONE — New NED Synclavier System

□ **SYNTONE** (Boston) expanded its synthesizer capabilities with the acquisition of a 32-voice, stereo **New England Digital** Synclavier Digital Music System with guitar and sampling options, a **Fostex B-16D** with autolocator, an **Allen and Heath CMC-24 semi-automated conole, and an Otari** two-track machine. Outboard gear includes a **Yamaha** digital reverb, and a **Lexicon PCM-42** digital delay line. In addition, the facility has also built an isolation booth. *1108 Boylston Street, #302, Boston, MA 02115. (617) 267-4137.*

□ **MIXMASTERS** (San Diego) is a 24-track facility geared for film scoring, video post production, and album sessions. According to owner **Charles Defazio**, studio equipment comprises: a **Neotek Series IIIc** 36-in/32-out console; various **Otari** tape machines, including an **MTR-90** 16/24-track, **Mk IIIA** half-inch eight-track, **MTR-12** and **MTR-10** two-track mastering machines; and outboard gear that includes **Lexicon Model 224** and **Model 200** reverbs, plus **Prime Time** and **Model 95 Prime Time II** effects processors, **Eventide Harmonizer**, and **Aphex Aural Exciter**. In addition, **Louie Stevens** and **Alan Harper** have been appointed as general manager/chief engineer and studio manager/engineer, respectively. *4877 Mercury Street, San Diego, CA 92111. (619) 569-7367.*

□ **ESPN** (Bristol, CT) has acquired a second 24-channel **Neve 5114** with 12 stereo and 12 mono busses. **Mike Negri**, director of engineering and maintenance, reports that the desk was purchased primarily for use on the network's *Sportscenter* program, which profiles highlights from professional and college sports. In addition, a **Sony 5000** video editor was added to the newly completed timecode editing suite. *ESPN Plaza, 936 Middle Street, Bristol, CT 06010. (203) 584-8477.*

Midwest:

□ **JOR-DAN STUDIO** (Wheaton, IL) has added a complete **AMS RMX-16** reverberation system, and microphones from **Neumann**, **AKG**, and **Beyer**. Also completed is a 600-foot musician's lounge with monitoring facilities. The facility was originally designed by **John Edward** and **George Augspurger**. *100 Wheaton Oaks Court, Wheaton, IL 60187-3043. (312) 653-1919.*

JOR-DAN — AMS reverb acquisition

□ **CHARLES BROWN MUSIC** (Cincinatti) has opened two new studios. The Palm Room is a SMPTE interlock facility that features an **Allen and Heath Syncon B** console, **Otari MX-5050** eight-track, **MTR-12** two-track (with SMPTE timecode center-track), and a specially modified **MTR-12** four-track for audio-visual work. The second studio, a MIDI-controlled Synth Room, boasts a **Yamaha DX7, QX1**, and **TX816 system with a Roland SBX-80 Sync Box** for SMPTE timecode interlock. Other equipment includes two **Oberheim Xpanders**, **LinnDrum**, **Chroma Polaris** synthesizer, and **J.L. Cooper** MIDI patch bay. Recently added to the 24-track Crimson Room and to both new rooms was a video-switching network linking the entire complex to an in-house one-inch video editing suite. *1349 East McMillan Avenue, Cincinnati, OH 45206. (513) 281-5212.*

□ **POGO RECORDS RECORDING STUDIO**, (Champaigne, IL) is a new 16-track recording facility featuring a **Studiomaster** 24-by-eight mixing console, a **UREI** tube console, a **3M** 16-track, a **Lexicon** digital reverb, and **Altec 604E** monitors. The studio, designed by **Combo Audio**, measures 500 square feet, with a 270 square-foot control room. The facility is an independent recording studio, and was designed by **Mark Rubel** (*pictured here*) within 75-year-old brick building. *37 East Taylor Street, Champaigne, IL 61820. (217) 351-8155.*

Southeast:

□ **AIRSHOW, INC.**, (Arlington, VA) has added the following equipment to enhance its new production studio and on-location remote recording capabilities: an **Ampex ATR-102** two-track; a **Bryston 2B-LP** amplifier; **Yamaha NS-10** close-field speakers; two **Lang PEQ** equalizers; **Trompeter WE**-style coaxial patchbays for digital audio and video signal routing; and a **Panasonic** video monitor with pulse cross and underscan display. *5727 25th Road North, Arlington, VA 22207. (703) 237-8312.*

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STUDIO UPDATE

FACILITIES
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☐ **SOUNDSCAPE STUDIOS** (Atlanta) is a new 24-track facility comprising a custom **Neotek Series IIIc** 28-by-24 mixing desk, with stereo submasters and group muting; various **Studer** tape machines, including an **A80 MkIII** 24-track, **A80** half-inch, **A810** quarter-inch and **A710** cassette deck; **FM Acoustics 800A** and **300A**, and **Hafler 500** monitors amps; plus **Tannoy** reference speakers. Outboard gear includes a **Lexicon 224X** digital reverb with **LARC**, **Super Prime Time** digital delay, and a **PCM-41** DDL, **Eventide H910** and **H949 Harmonizers**, **dbx Model 165A** limiters; **Valley People Kepex II** noise gates; and an **Audio Arts 4100** equalizer. *677 Antone Street NW, Atlanta, GA 30318. (404) 351-1003.*

Southern California:

☐ **KDISC** (Hollywood) has purchased a **Sony** digital editing and disk-mastering audio system consisting of a **PCM-1610** processor, **DDU-120** digital delay, a pair of **BVU-800 db** recorders, and a **DAE-1100** editing system. Pictured here are **Ken Perry** (left) and **John Golden** with the new system. Currently, the facility claims to offer a full range of in-house analog and digital disk-mastering services to analog disk, digital and/or analog tape, music editing and assembly for Compact Disc release. The Sony system augments Studio A's exisiting tandem **Neumann VMS 80** analog disk-cutting lathes, and **Zuma** computer equipped **Neumann VMS 70** lathe with **Technics** quartz drive motor located in Studio B. *6550 Sunset, Hollywood, CA 90028. (213) 466-1323.*

KDISC — Sony digital editing system

☐ **TIM JORDAN RENTALS** (Los Angeles) has purchased six **Timeline Lynx** timecode modules, and says it is the first company to be offering the synchronization units for rent on the West Coast. The Lynx modules consist of individual timecode generator/reader/resolver units easily connected to virtually all digital and analog video, audio, and film transports. In addition, the modules can be linked together for up to a 32-machine synchronization. According to facility owner **Tim Jordan**: "I no longer have to stock a variety of interface circuit boards and EPROM sets for the many different recorders, and there is no prior set up time necessary at my shop [when renting the Lynx]." *8474 West Third Street, Los Angeles, CA 90048 (213) 653-0240.*

☐ **OCEANWAY RECORDERS** (Hollywood) has installed a second **GML Moving Fader Automation System** onto a custom 32-input Delcon console, which is linked to a portable 16-input API board, resulting in a total of 48 channels of automation for Studio B. The dual console system, which tailors mainly to album projects and film scores, was designed by **Jay Kaufman** of Oceanway. The facility's first GML system was installed a year ago onto an API desk. A 40-input GML system was also delivered to **Mama Joe's**, in North Hollywood, and installed onto a **Trident Series 80** board. With these two systems, there now are reported to have been six GML systems installed this year. *6050 Sunset, Hollywood, CA 90028. (213) 467-9375.*

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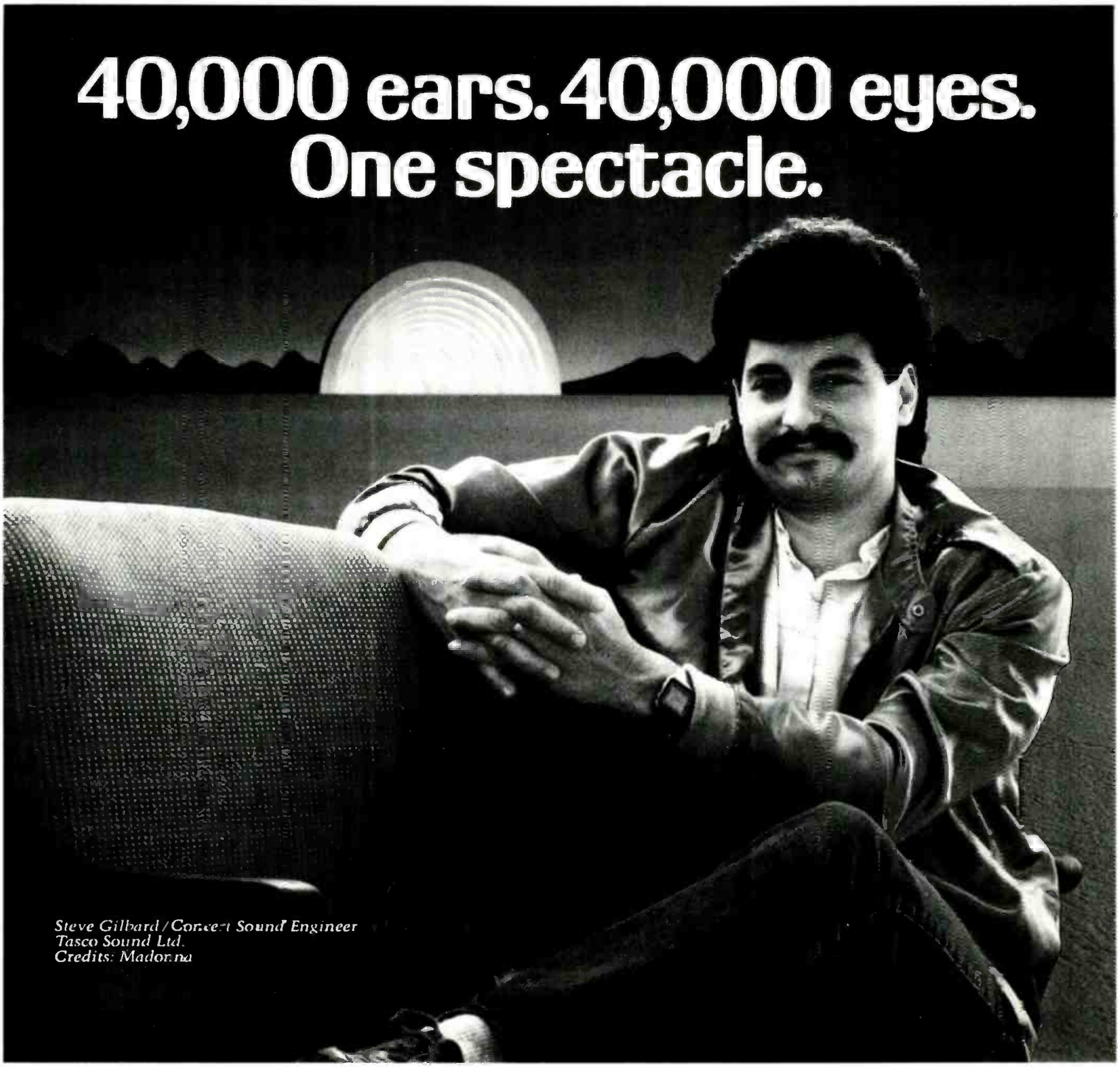

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P.O. Box 1511
Cincinnati, OH 45202
(513) 621-6389 DM, TD, PR, PL, PK

STUDIO PRESSING SERVICE
320 Mill St.
Lockland, OH 45215
(513) 793-4944 TD, DM, PR, PL

SUMA RECORDING STUDIO
5706 Vrooman Rd.
Cleveland, OH 44077
(216) 951-3955 DM, TD, PL, PR, PK

TRIAD PRODUCTIONS
1910 Ingersoll Ave.
Des Moines, IA 50309
(515) 243-2125 TD

Mountain:

BONNEVILLE MEDIA COMMUNICATIONS
130 Social Hall Ave.
Salt Lake City, UT 84111
(801) 237-2677 TD

CHRISTIAN AUDIO TAPES
3005 W. Glendale Ave.
Phoenix, AZ 85021
(602) 246-4976 TD

MIRROR IMAGE
10288 E. Jewell Ave., #45
Denver, CO 80231
(303) 751-2268 TD

MOONVALLEY CASSETTE
10802 N. 23rd Ave.
Phoenix, AZ 85029
(602) 864-1980 TD, PK

RAINBOW CASSETTE STUDIO
P.O. Box 472
Taos, NM 87571
(505) 776-2268 TD

RAINBOW VENTURES STUDIO
2219 W. 32nd Ave.
Denver, CO 80211
(303) 433-7231 DM, TD, PL, PR, PK

ROCKY MOUNTAIN RECORDING
8305 Christensen Rd.
Cheyenne, WY 82009
(307) 638-8733 DM, PL, PR

SOUNDMARK, LTD.
4950-C Nome St.
Denver, CO 80239
(303) 371-3076 TC

TALKING MACHINE
6733 N. Black Canyon Highway
Phoenix, AZ 85015
(602) 246-4238 DM, PK
- Ortofon Cutting System
- Zuma Computer-Controlled Lathe
- Consumer Digital Formats Transferred Direct to Disk VHS, Beta, U-Matic Video

UNIVERSAL AUDIO SALES CORP.
6540 East Lafayette Blvd.
Scottsdale, AZ 85251
(602) 994-5528 TD, PM, PK

Southern California:

ABBEY TAPE DUPLICATORS, INC.
9525 Vassar Ave.
Chatsworth, CA 91311
(818) 882-5210 TD, PK

ALLIED RECORD CO., WEA MFG.
6110 Peachtree St.
Los Angeles, CA 90040
(213) 725-6900

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ALSHIRE INTERNATIONAL, INC.
1015 Isabel St., P.O. Box 7107
Burbank, CA 91510
(213) 849-4671 DM, TD, PL, PR, PK

ARTISAN SOUND RECORDERS
1600 N. Wilcox Ave.
Hollywood, CA 90028
(213) 461-2751 DM

AUDIO CASSETTE DUPLICATORS
5816 Lankershim Blvd., #7
North Hollywood, CA 91601
(818) 762-2232 TD

AUDIO VIDEO CRAFT, INC.
7000 Santa Monica Blvd.
Los Angeles, CA 90038
(213) 466-6475 TD

AWARD RECORD MFG., INC.
5200 W. 83rd St.
Los Angeles, CA 90045
(213) 645-2281 DM, TD, PL, PR, PK

BAMCO RECORDS
1400 S. Citrus Ave.
Fullerton, CA 92633
(714) 738-4257 PR

BUZZY'S RECORDING SERVICES
6900 Melrose Ave.
Los Angeles, CA 90038
(213) 931-1867 TO

CMS DIGITAL RENTALS, INC.
453-E Wapello St.
Altadena, CA 91001
(818) 797-3046 CO

CAPITOL RECORDS STUDIOS
1750 N. Vine St.
Hollywood, CA 90028
(213) 462-6252 OM, TD

CASSETTE PRODUCTIONS UNLIMITED
46 S. DeLacey St. Suite 24
Pasadena, CA 91105
(818) 449-0893

CUSTOM DUPLICATING, INC.
3404 Century Blvd.
Ingelwood CA 90303
(213) 670-5575 TD, PK

DYNASTY STUDIO
1614 Cabrillo Ave.
Torrance, CA 90501
(213) 328-6836 TD

FILAM NATIONAL PLASTICS, INC.
13984 S. Orange Ave.
Paramount, CA 90723
(213) 630-2500 PK

BERNIE GRUNDMAN MASTERING
6054 Sunset Blvd.
Hollywood, CA 90028
(213) 465-6264 CD, DM

HITSVILLE STUDIOS
7317 Romaine St.
Los Angeles, CA 90046
(213) 850-1510 DM, CD

JVC CUTTING CENTER
6363 Sunset Blvd., #500
Hollywood, CA 90028
(213) 467-1166 DM, CD

K-DISC
26000 Spring Brook Ave.
Saugus, CA 91350
(805) 259-2360 DM, PL, PR, PK

K.M. RECORDS
2980 N. Ontario St.
Burbank, CA 91504
(818) 841-3400 DM, PL, PR, PK

LIGHTNING CORP.
7854 Ronson Rd.
San Diego, CA 92111
(619) 565-6494 TD, PK

LOCATION RECORDING
2201 W. Burbank Blvd.
Burband, CA 91506
(818) 849-1321 DM, PK, PR, PL

MCA WHITNEY RECOROING STUDIO
1516 W. Glenoaks Blvd.
Glendale, CA 91201
(213) 245-6801 DM, TO

ML TAPE DUPLICATING
6935 Valican
Van Nuys, CA 91406
(818) 988-2737 TO

MASTER DIGITAL, INC.
1749 14th St.
Santa Monica, CA 90404
(213) 452-1511 TD

MOBILE FIDELITY SOUND LAB
P.O. Box 4285
Chatsworth, CA 91313-4285
(818) 709-8440 CD, TD, DM

MONARCH RECORD PRESSING
9545 San Fernando Rd.
Sun Valley, CA 91352
(818) 767-8833 TD, PR, PL, PK

OPHARION RECORDINGS
P.O. Box 91209
Long Beach, CA 90809
(213) 438-4271 TD

PRECISION LACQUER
1008 North Cole Ave.
Hollywood, CA 90038
(213) 464-1008 DM

PRESENT TIME RECORDERS
5154 Vineland Ave.
North Hollywood, CA 91601
(818) 762-5474 TD

QUAD TECK STUDIOS AND F.D.S. LABS
4007 West 6th St.
Los Angeles, CA 90020
(213) 383-2155 TD, DM

RAINBO RECORD MFG. CORP.
1738 Berkeley St.
Santa Monica, CA 90404
(213) 829-0355 DM, TD, PL, PR, PK

RECORD TECHNOLOGY, INC.
486 Dawson Drive
Camarillo, CA 93010
(805) 484-2747 TD, PL, PR

SHEFFIELD LAB MATRIX
1830 Olympic Blvd.
Santa Monica, CA 90404
(213) 829-7825 PL

SOUND MASTER AUDIO/VIDEO
10747 Magnolia Blvd.
North Hollywood, CA 91601
(213) 650-8000 DM

STUDIO MASTERS
8312 Beverly Blvd.
Los Angeles, CA 90048
(213) 653-1988 DM

TAKEDA RECORD SERVICE
11542 Burbank Blvd. #2
North Hollywood, CA 91601
(818) 760-6644 TD, DM, PR, PL, PK

TAPE SPECIALTY, INC.
13411 Saticoy St.
North Hollywood, CA 91605
(818) 786-6111 TD, PK

TAPELOG
10511 Keokuk Ave.
Chatsworth, CA 91311
(818) 882-4433 TD

VIRCO RECORDING, INC.
700 S. Date Ave.
Alhambra, CA 91803
(213) 283-1888 DM, TD, PL, PR

Northern California:

ARCAL
2732 Bay Road
Redwood City, CA 94063
(415) 369-7348 TD

AUDIODYNE
P.O. Box 825
San Jose, CA 95106
(408) 287-3520 TD

KENNETH BACON ASSOC., INC.
24 Commercial Blvd., Suite E
Novato, CA 94947
(415) 883-5041 TD, PK

DAVKORE COMPANY
1300-D Space Park Way
Mountain View, CA 94043
(415) 969-3030 TD, PK

FANTASY STUDIOS
10th and Parker
Berkeley, CA 94710
(415) 549-2500 DM

BILL RASE PRODUCTIONS, INC.
955 Venture Court
Sacramento, CA 95825
(916) 929-9181 TD, PR

SHUR-SOUND & SIGHT, INC.
3350 Scott Blvd. #5
Santa Clara, CA 95054
(408) 727-7620 TD

SONIC ARTS/MASTERING ROOM
665 Harrison St.
San Francisco, CA 94107
(415) 781-6306 TD, CD, TM, DM, PR, PL, PK

Northwest:

AMERICAN TAPE DUPLICATING
7017 15th Ave. N.W.
Seattle, WA 98117
(206) 789-8273 TD, PK

AUDIO PRODUCTION STUDIO
7404 Sand Lake Road
Anchorage, AK 99502
(907) 243-4115 TD, PR, PK

CAPITOL CITY STUDIOS
911 East 4th Ave.
Olympia, WA 98506
(206) 352-9097 TD, DM, PR, PL, PK

CASSETTE TECHNOLOGIES
5722 Swan Creek Drive, E.
Tacoma, WA 98404
(206)472-2740 TD, PK

NORTHWEST, INC.
1224 S.W. Broadway
Portland, OR 97205
(503) 226-0170 TD

Canada:

McCLEAR PLACE MASTERING STUDIOS
225 Mutual St.
Toronto, Ontario M5B 2B4
(416) 977-9740 DM

WORLD RECORDS
P.O. Box 2000 Baseline Road West
Bowmanville, Ontario L1C 3Z3
1(800) 263-7798 DM, TD, PL, PR, PK

Foreign:

The Directory

R-e/p's Product Listing of TIME DOMAIN PROCESSORS and SPECIAL EFFECTS UNITS

Coming in the next issue: Frequency and Dynamics Processors.

ACOUSTICLOG, INC.
19 Mercer Street
New York, NY 10013
Phone: (212) 925-1365

Phase 5A

Inputs: One.
Outputs: One.
Effects Type(s): Phase shifting.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Manual sweep; sweep speed; intensity; regeneration; and envelope follower level.
Selected Standard Features: Built-in regeneration and envelope follower; CV input and envelope follower output for connection to other devices.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: N/A
For additional information circle #170

ADVANCED MUSIC SYSTEMS
U.S. Distributor: Harris Sound, Inc.
6640 Sunset Blvd. Suite #110
Hollywood, CA 90028
Phone: (213) 469-3500

AMS DMX 15-80S

Inputs: Two.
Outputs: Two.
Effects Type(s): Delay, pitch-change, and sampling.
Delay Ranges: 0 to 13 seconds (full bandwidth).
Reverb/Echo Parameters: N/A.
Operational Controls: Two-channel input-output; regeneration; VCO (speed and depth); nudge buttons; and keypad for adding delay.
Selected Standard Features: Channel A has 3.2 seconds with pitch-change and sampling; channel B has 1.6 seconds and pitch-change.
Frequency Response (input/output): 20 Hz to 18 kHz, -3/+0 dB.
Distortion: THD less than 0.03% at 1 kHz full output.
S/N Ratio (input/output): 90 dB dynamic range.
Pro-User Price Range: $8,995

AMS RMX-16

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb with special effects.
Delay Ranges: Delay program: 0 to 1.2 seconds.
Reverb/Echo Parameters: Pre-delay; decay time; decay filters (low and high).
Operational Controls: Nudge buttons and keypad for changing reverb parameters and programs; store control (save and recall); nine stores; and input/output pots.
Selected Standard Features: 12 factory programs; three programs erasable and reprogrammable via optional remote with barcode reader.
Frequency Response (input/output): 20 Hz to 18 kHz, -3/+0 dB.
Distortion: THD less than 0.03% at 1 kHz full output.
S/N Ratio (input/output): 90 dB dynamic range.
Pro-User Price Range: $7,450
For additional information circle #171

AKG ACOUSTICS, INC.
77 Selleck Street
Stamford, CT 06902
Phone: (203) 348-2121

Model TDU7000

Inputs: Four.
Outputs: Four (standard model).
Effects Type(s): Digital delay.
Delay Ranges: 0 to 299 ms, extendable in 800 ms per module.
Reverb/Echo Parameters: N/A.
Operational Controls: Adjustable inputoutput levels; and delay thumbwheels.
Selected Standard Features: Input and output modules configurable and expandable; optional effects module.
Frequency Response (input/output): 20 Hz to 16 kHz.
Distortion: Less than 0.2% for 20 Hz to 6 kHz.
S/N Ratio (input/output): Better than -93 dB, A weighted.
Pro-User Price Range: For standard unit: $15,000

Model BX25E

Inputs: Two.
Outputs: Two (standard model).
Effects Type(s): Spring reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: 1.5 to 3.5 seconds.
Operational Controls: Remote control mix of echo and reverb, as well as reverb types.
Selected Standard Features: Input and output levels adjustable, with optional M250 echo unit at 60 ms selectable (two per channel).
Frequency Response (input/output): 50 Hz to 8 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -76 dB, A weighted.
Pro-User Price Range: For standard unit: $5,500
For additional information circle #172

ALESIS CORP.
P.O. Box 3908
Los Angeles, CA 90078
Phone: (213) 467-8000

Model XT

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverberation.
Delay Ranges: Decay time range is 0.15 to 10 seconds.

Reverb/Echo Parameters: Pre-delay; slap back; diffusion; size; decay time; and filters.
Operational Controls: Input/output level; mix; defeat; external defeat.
Selected Standard Features: 16-bit audio.
Frequency Response (input/output): 20 kHz dry; 14 kHz reverb.
Distortion: Less than 0.1%.
S/N Ratio (input/output): Dynamic range is less than or equal to -85 dB; typical is less than or equal to -92 dB.
Pro-User Price Range: $795 and up
For additional information circle #173

APPLIED RESEARCH AND TECHNOLOGIES, INC.
215 Tremont Street
Rochester, NY 14608
Phone: (216) 436-2220

Model DR1-240

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital reverb system.
Delay Ranges: N/A.
Reverb/Echo Parameters: Seven.
Operational Controls: Room type; pre-delay; HF damping; blend; decay time; and diffusion.
Selected Standard Features: Remote control; MIDI; stereo in; stereo out; and 100 presets.
Frequency Response (input/output): 10 Hz to 14 kHz.
Distortion: Less that 0.1%.
S/N Ratio (input/output): Better than -84 dB.
Pro-User Price Range: $1,695

Model 01A-191

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb system.
Delay Ranges: N/A.
Reverb/Echo Parameters: Seven.
Operational Controls: Room type; pre-delay; HF damping; blend; decay time; and diffusion.
Selected Standard Features: 49 presets; nine room types; balanced mono in; and stereo out.
Frequency Response (input/output): 20 Hz to 10 kHz.
Distortion: Less that 0.025%.
S/N Ratio (input/output): Better than -72 dB.
Pro-User Price Range: $1,395

Model DR2-230

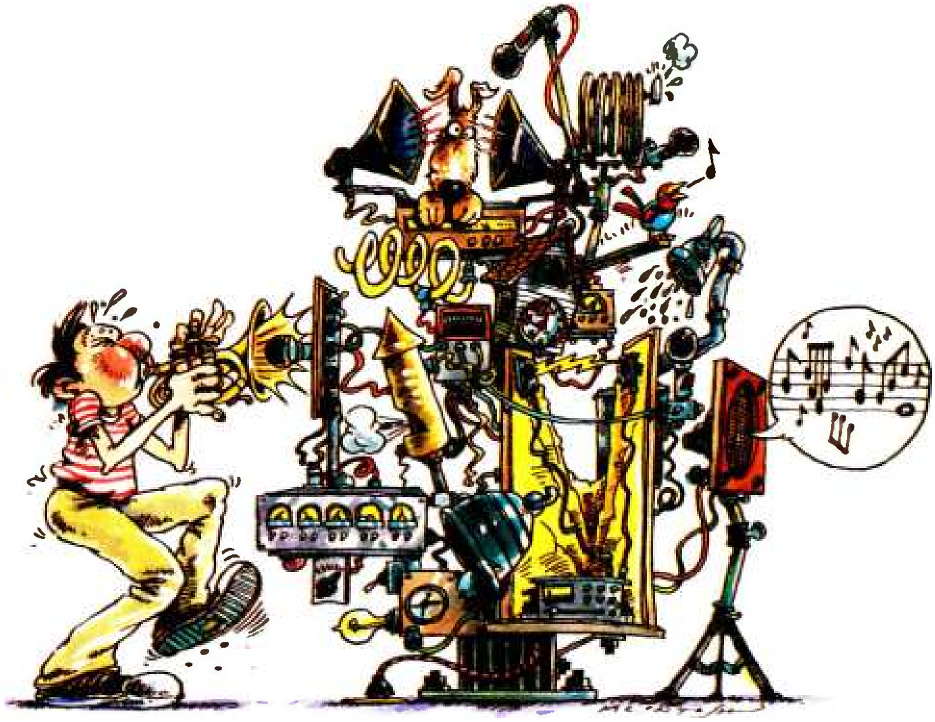
Inputs: One.
Outputs: Three.
Effects Type(s): Digital reverb system.
Delay Ranges: N/A.
Reverb/Echo Parameters: Six.
Operational Controls: Room type; pre-delay; HF damping; blend; decay time; and diffusion.
Selected Standard Features: Nine room types; three presets; bypass; balanced mono in/stereo or mixed mono out; and Hi/Lo level.
Frequency Response (input/output): 20 Hz to 10 kHz.
Distortion: Less that 0.025%.
S/N Ratio (input/output): Better than -72 dB.
Pro-User Price Range: $995

Model 1500

Inputs: One.
Outputs: One.
Effects Type(s): Digital delay.
Delay Ranges: 0.15 ms to 1.5 seconds in four ranges.
Reverb/Echo Parameters: N/A.
Operational Controls: Manual; width; speed; mix; and regeneration.
Selected Standard Features: Repeat hold; in/out; and Hi/Lo level.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: Less that 0.2%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $500

Models 129/130/131

Inputs: One.
Outputs: One.
Effects Type(s): Pitch transpose, display, and foot pedal in one unit.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Pitch shift: octave up, octave down; remote foot control; pitch display readout; and bypass.
Selected Standard Features: Four presets; mix; regeneration; instrument/line inputs; and auxiliary loops.
Frequency Response (input/output): 15 Hz to 11 kHz.
Distortion: 0.25%.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $1,200
For additional information circle #174



HARRY, THERE *HAS* TO BE AN EASIER WAY.

MEMO:

Listen, Harry, I *know* you keep saying we need "creative sound processing" to stay competitive. I *loved* the way you hung the mikes inside a 24-gallon aquarium for the Fred's Fish Food jingle (too bad Fred's singing goldfish dropped dead, though). And your reverse hyperspatial time-delay effects for the "H.G. Wells Concerto" were *incredibly brilliant*. Real award-winning stuff.

But I gotta tell you: these complicated setups of yours are driving me crazy. First I spend *all day* rigging equipment. Then I go *all night* de-bugging the effects so they sound right.

Harry, there just *has* to be an easier way to produce interesting acoustic environments.

And I think I found it: Ursa Major's new *StarGate 626*. The 626 puts just about every effect we need—digital reverb, delays, and special effects—inside *one box* with *one* set of controls. The reverb programs all sound *absolutely professional* (this is an Ursa Major unit, after all)—but the 626 goes way beyond straight reverb. There's mono and stereo delay lines, for example, an effect called "reverse reverb," a stereoized dual echo, and the brightest plate simulation I've ever heard. Plus a lot more—16 pre-tuned "rooms" in all, with 256 possible *variations* on each effect.

Anyway, Harry, I want you to cancel everything on your calendar tomorrow morning. I'm taking you to hear a *live demo* of the 626. Don't forget the checkbook, either. We *need* this thing—and the sooner the better.

Regards,
Ozzie

THE STARGATE 626

URSA MAJOR, Inc.

Box 28, Boston, MA 02258 USA • Telephone (617) 924-7697
Telex: 921405 URSAMAJORBELM

For additional information circle #79

The Directory

AUDIO+DESIGN CALREC, INC.
P.O. Box 786
Bremerton, WA 98310
Phone: (206) 275-5009

Panscan

Inputs: Three.
Outputs: Two.
Effects Type(s): Auto panning and beat counter.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Pan speed; pan depth; image offset; auto external trigger; image freeze; manual trigger; beat count; and beat count threshold.

Selected Standard Features: Beat counter from 1 to 10 beats; and rapid vibrato for soft shift.
Frequency Response (input/output): 20 Hz to 20 kHz, +0/-1 dB.
S/N Ratio (input/output): Better than -88 dB, reference +8 dBm.
Pro-User Price Range: $890

SCAMP S23

Inputs: Two.
Outputs: Two.
Effects Type(s): Auto panning.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Pan speed; pan pattern; rate of hold off (delay) between pans.
Selected Standard Features: External trigger (key) input; triggered or continuous pan; and normal/reverse indicators.
Frequency Response (input/output): 20 Hz to 20 kHz, ±0.05 dB.
Distortion: Less than 0.05%.
S/N Ratio (input/output): Better than -103 dB, reference +8 dBm.
Pro-User Price Range: $450

SCAMP S24

Inputs: Two.
Outputs: Two.
Effects Type(s): ADT/flange.
Delay Ranges: Two ranges: 1.2 to 11 ms; 11 to 45 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Positive/negative flanging; envelope follower; oscillator frequency and modulation; and spin (feedback).
Selected Standard Features: Front-end limiting allows controlled "100%" feedback; stereo reverb by cross coupling two units; and additional delay with two or more units.
Frequency Response (input/output): 20 Hz to 17 kHz, ±0.05 dB.
Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -80 dB, reference +8 dBm.
Pro-User Price Range: $650
For additional information circle #175

AUDIO/DIGITAL, INC.
1000 S. Bertelsen Road, Suite #4
Eugene, OR 97402
Phone: (503) 687-8412

ADX 2000

Inputs: Six.
Outputs: 64.
Effects Type(s): Cluster alignment; speaker alignment; theme parks; and touring sound.
Delay Ranges: 10 microseconds to 1.048 seconds.
Reverb/Echo Parameters: Delay; gain; and on/off control.
Operational Controls: RS232 port; user memory; and front-panel keypad.
Selected Standard Features: Microprocessor controlled; easily programmed; saves several delay configurations in memory; and security lock-out.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: 0.05% THD plus noise at 1 kHz.
S/N Ratio (input/output): Better than -105 dB.
Pro-User Price Range: For one-in/four-out model: $3,500
For additional information circle #176

ADA SIGNAL PROCESSORS
7303D Edgewater Drive
Oakland, CA 94621-3095
Phone: (415) 632-1323

Digitizer 4

Inputs: One.
Outputs: Two.
Effects Type(s): Programmable digital delay.
Delay Ranges: 1 to 1.024 seconds.

Reverb/Echo Parameters: N/A.
Operational Controls: Full programming of all decay functions.
Selected Standard Features: 16 user-programs; stereo outputs; 16 factory shadow programs; self-diagnostics; and LED readout.
Frequency Response (input/output): Decay: 20 Hz to 16 kHz.
Distortion: 0.02% maximum at 0 dBv, wet.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $699.95

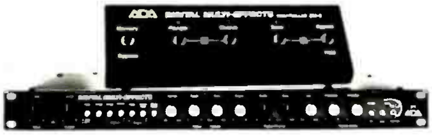
2FX Digital Multi-Effects

Inputs: One.
Outputs: Two.
Effects Type(s): Delay/effects.
Delay Ranges: N/A.

For additional information circle #80

The Directory

Reverb/Echo Parameters: 10 to 1 modulation depth.
Operational Controls: Separate controls for flange, chorus and echo.



Selected Standard Features: User can run two delays at once; patch reverse switch; delay rate LED; and optional foot controller.
Frequency Response (input/output): Wet: 20 Hz to 17 kHz.
Distortion: 0.5% maximum at 0 dBv, wet.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $599.95

64i/1.28i/2.56i Digital Delay

Inputs: One.
Outputs: Two.
Effects Type(s): Full-function digital delay.

Delay Ranges: 0.31 to 640 ms; 0.15 to 1.280 seconds; 0.31 to 2.560 seconds, respectively.
Reverb/Echo Parameters: 8-to-1 modulation depth.
Operational Controls: Input and output levels; delay/dry mix; full modulation section; phase invert; and repeat hold.
Selected Standard Features: Variable hi-cut for regeneration; modulation waveform control; delay rate LED; modulation speed LEDs; and four-step LED headroom indicator.
Frequency Response (input/output): Wet: 20 Hz to 17 kHz.
Distortion: 0.5% maximum at 0 dBv, wet.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: 64i: $499.95; 1.28i: $699.95; 2.56i: $799.95

Model STD-1

Inputs: One.
Outputs: Two.
Effects Type(s): Multi-tap delay line.
Delay Ranges: 1.3 to 55.5 ms.
Reverb/Echo Parameters: Six, non-harmonically related delay taps for stereo and psycho-acoustic processing.
Operational Controls: N/A.
Selected Standard Features: True stereo outputs; six assignable delay taps; and full modulation section for multi-voice chorus.
Frequency Response (input/output): Dry: 10 Hz to 20 kHz; delay: 20 Hz to 13.5 kHz.
Distortion: 0.55% maximum at 0 dBv, wet.
S/N Ratio (input/output): Better than -93 dB unweighted.
Pro-User Price Range: $799.95

Model S1000 Digital Delay

Inputs: One.
Outputs: Two.
Effects Type(s): Full-function digital delay.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Input/output levels; delay/dry mix; full modulation section; phase invert; and repeat hold.
Selected Standard Features: Four-step LED headroom indicator; and 8-to-1 modulation depth.
Frequency Response (input/output): Delay: 20 Hz to 10 kHz.
Distortion: 0.5% maximum at 0 dBv, wet.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $299.95
For additional information circle #177

BARCUS-BERRY ELECTRONICS
5381 Production Drive
Huntington Beach, CA 92649
Phone: (714) 898-9211

Model BBE 202

Inputs: One.
Outputs: One.
Effects Type(s): Phase and amplitude distortion eliminator.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Stereo/mono; process-bypass; amplitude/gain; 50 Hz adjustable; and level.
Selected Standard Features: XLR input and output.
Frequency Response (input/output): 5 Hz to 20 kHz, ±0.1 dB.
Distortion: Less than 0.1% at 1 kHz.
S/N Ratio (input/output): N/A.
Pro-User Price Range: $1,495
For additional information circle #178

BEL ELECTRONICS
UK Distribution: Studio Equipment Distributors
29 Guilford Street, 29th Floor
Luton, Bedfordshire
LU1 2NQ, England
Phone: (0582) 452495

Models BD80/240/320

Inputs: One.
Outputs: Three.
Effects Type(s): Digital delay processors.
Delay Ranges: 80: Four seconds maximum delay; 240: 12 seconds maximum delay; 320: 16 seconds maximum delay.
Reverb/Echo Parameters: N/A.
Operational Controls: Feedback control with phase invert switch and filter switch; audio trigger; repeat hold; truncation; and two operating modes: manual (via up/down keypad switches) and oscillator (via speed/depth controls).
Selected Standard Features: LED headroom indicator; LED indicated infinite repeat switch; output mix control; stereo input and mono outputs on jacks; XLR connectors optional.
Frequency Response (input/output): (For manual mode) 80: 20 Hz to 15 kHz; 240: 20 Hz to 18 kHz; 320: 20 Hz to 15 kHz.
Distortion: 0.2% maximum at 1 kHz.
S/N Ratio (input/output): Dynamic range: 85 dB typical.
Pro-User Price Range: N/A
For additional information circle #179

BIAMP SYSTEMS
P.O Box 2160
Portland, OR 97208
Phone: (503) 641-6767

MR140/SR340

Inputs: One.
Outputs: One.
Effects Type(s): Spring reverb system.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Input and output levels, wet dry, and four bands of EQ.
Selected Standard Features: Footswitch jack; plus two pre-spring limiters.
Frequency Response (input/output): N/A.
Distortion: THD less than 0.01%.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: MR140: $349; SR340: $495
For additional information circle #180

BOSS
U.S. Distributor: Roland Corp.
7200 Dominion Circle
Los Angeles, CA 90040
Phone: (213) 685-5141

RDD-10 DDL

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital delay.

Delay Ranges: 0 to 400 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Delay range; fine rate/-depth modulation (±0.5 to 1.5); feedback; and delay tone/level.
Selected Standard Features: RCA inputs/outputs; stereo modulation capabilities.
Frequency Response (input/output): 40 Hz to 16 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: $275

CE-300 Super Chorus

Inputs: One.
Outputs: Two.
Effects Type(s): Stereo chorus.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Input gain; modulation (rate and depth); tone; output; bypass; and direct mute.
Selected Standard Features: +12 dB input level; and two separate stereo choruses.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: $299

RX-100 Reverb Box

Inputs: Three.
Outputs: Four.
Effects Type(s): Reverb unit.
Delay Ranges: 20 to 400 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Input gain switch; channel A and B volume; reverb volume; panpot; and mode switch.
Selected Standard Features: Stereo capability; mode one allows phase cancelation of 180 degrees.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: $230

RBF-10 Flanger

Inputs: Two.
Outputs: Two.
Effects Type(s): Flanger.
Delay Ranges: 1 to 13 ms.
Reverb/Echo Parameters: 100 ms to 16 seconds LFO.
Operational Controls: Manual; rate; depth; level; balance; and effect on/off.
Selected Standard Features: RCA or jack connection; stereo capability.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: $160

RPH-10 Phaser

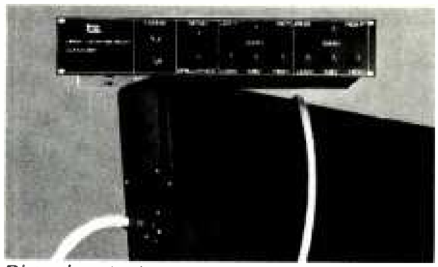
Inputs: Two.
Outputs: Two.
Effects Type(s): Phase shifter.
Delay Ranges: LFO rate, 100 ms to 14 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Mode I, II, III; manual; rate; depth; and resonance.
Selected Standard Features: Two-stage phasing; RCA or jack connection; stereo capable.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $160
For additional information circle #181

BRICK AUDIO
102 South Porter
Elgin, IL 60120
Phone: (312) 742-7425

Plate Controller

Inputs: One.
Outputs: Two.
Effects Type(s): Plate reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: 0.5 to 7 seconds, depending on size ordered.
Operational Controls: N/A.
Selected Standard Features: "Brilliance" on send; three-band EQ on each return; plate controller is a rack mounting, plate chamber is four inches deep.

Frequency Response (input/output): 100 Hz to 20 kHz, ±4 dB.



Distortion: N/A.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $995 and up
For additional information circle #182

CONNECTRONICS CORP.
652 Glenbrook Road
Stamford, CT 06906
Phone: (203) 324-2889

Accessit and GBS Reverbs

Inputs: One and two respectively.
Outputs: Two.
Effects Type(s): Spring Reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: 2.5 seconds.
Operational Controls: Accessit: Mix; input match; and sweep EQ; GBS: input match.
Selected Standard Features: Accessit: Outboard spring tank.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): N/A.
Pro-User Price Range: Accessit: $349; GBS: $715
For additional information circle #183

DELTA LAB/ADS
One Progress Way
Wilmington, MA 01887
Phone: (617) 658-5100

CE 1700 NVM Compu Effectron

Inputs: One.
Outputs: One.
Effects Type(s): Programmable digital delay

For additional information circle #83

The Directory

processor.
Delay Ranges: 0 to 1.724 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Phase; input; feedback; delay; sweep; ratio; mix; output; store; recall; setup; repeat; record; play; bypass; and numeric keypad.
Selected Standard Features: Built in library of effects; sample and trigger mode; "glitch-free operation from one effect to the next"; real time processing; 219 user locations; remote capabilities; MIDI interface; and touch control front panel.
Frequency Response (input/output): 20 Hz to 20 kHz, all delay settings.
Distortion: 0.2%, maximum at 1 kHz.
S/N Ratio (input/output): Dynamic range 90 dB.
Pro-User Price Range: $999

Effectron III ADM 1030/
Effectron II ADM 1024/
Effectron I ADM 1020

Inputs: One.
Outputs: One.
Effects Type(s): Programmable digital delay processor.
Delay Ranges: 1030: 0.25 to 1.024 seconds; 1024: 0.22 to 1.024 seconds; 1020: 4 to 1.024 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Units include: input, positive/negative feedback, delay range, modulator, width, speed; 1030 includes: envelope, postive/negative mix, and write banks A/D.
Selected Standard Features: Programmable; 17 kHz bandwidth on all settings; all steel chassis; simulated stereo; optional footswitch for 1030 and 1024.
Frequency Response (input/output): 20 Hz to 17 kHz; 1020: 20 to 15 kHz.
Distortion: 0.2%, maximum.
S/N Ratio (input/output): Dynamic range 90 dB.
Pro-User Price Range: 1030: $699; 1024: $499; 1020: $329
For additional information circle #184

DOD ELECTRONICS, INC.
5639 Riley Lane
Salt Lake City, UT 84107
Phone: (801) 268-8400

Model R-848

Inputs: One.
Outputs: One.
Effects Type(s): Reverberation.
Delay Ranges: Pre-delay: 20 to 60 ms.
Reverb/Echo Parameters: Reverb time: 2 seconds.
Operational Controls: Pre-delay; drive level; reverb time; four contour presets; two-band quasi-parametric EQ; and mix output level.
Selected Standard Features: Variable pre-delay; frequency contour presets; footswitch controllable; and reverb kill switch.
Frequency Response (input/output): Wet: 5.6 kHz; dry: 20 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -72 dB.
Pro-User Price Range: $399.95

Model RDS-900/1900

Inputs: One.
Outputs: Three.
Effects Type(s): Digital flanging, chorus, doubling, and echo.
Delay Ranges: 1.5 to 14 ms; 6 to 55 ms; 50 to 470 ms; and 100 to 900 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Speed; width; delay time; repeat; hold; feedback; mix; output level; input level; and effect on/off.
Selected Standard Features: Footswitch controllable delay; kill and repeat hold functions; VCO control input; and dry output.
Frequency Response (input/output): Wet: 5.6 kHz; dry: 20 kHz.
Distortion: N/A.
S/N (input/output): Better than -85 dB wet; and -90 dB dry.
Pro-User Price Range: 900: $295.95; 1900: $299.95

Model RDS-3600

Inputs: One.
Outputs: Three.
Effects Type(s): Digital flanging, chorus, doubling, and echo.
Delay Ranges: 1.5 to 14 ms; 6 to 55 ms; 50 to 450 ms; 200 ms to 1.800 seconds, 400 ms to 3.600 seconds; and 800 ms to 7.200 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Speed; width; delay time; repeat; hold; feedback; mix; output level; input level; and effect on/off.
Selected Standard Features: Footswitch controllable delay; kill and repeat hold functions; VCO control input; and dry output.
Frequency Response (input/output): 40 to 15 kHz, at 1.800 seconds; 40 Hz to 7.5 kHz and 40 Hz to 3.7 kHz at extended delay ranges.
Distortion: N/A.
S/N Ratio (input/output): Better than -85 dB wet; and -90 dry.
Pro-User Price Range: $399.95
For additional information circle #185

EMT-FRANZ
U.S. Distributor: Gotham Audio Corp.
741 Washington Street
New York, NY 10014
Phone: (212) 741-7511

EMT 251/252

Inputs: One.
Outputs: Two.
Effects Type(s): Reverb; echo; chorus; and delay.
Delay Ranges: 0 to 440 ms.
Reverb/Echo Parameters: Variable, 0.4 to 18 seconds.
Operational Controls: Reverb time; reflection amplitude and time delay; and high, mid, and low reverb contour.
Selected Standard Features: 251: floor standing digital reverb with LCD parameter display; 252: rackmount processor with remote control panel and 128 memories.
Frequency Response (input/output): 30 Hz to 15 kHz.
Distortion: N/A.
S/N Ratio (input/output): 252: better than -75; 21 better than -73 dB.
Pro-User Price Range: 251/252: $16.5000; 252: $16.5000

EMT 245

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb.
Delay Ranges: 0 to 84 ms.
Reverb/Echo Parameters: Variable, 0.4 to 4.5 seconds.
Operational Controls: Dealy; reflection amplitude; reverb time; LF/HF reverb contour; and optional 10-memory remote control.
Selected Standard Features: Rackmount digital reverb.
Frequency Response (input/output): 30 Hz to 8 kHz.
Distortion: N/A.
S/N ratio (input/output): Better than -68 dB
Pro-User Price Range: $10.145

EMT 240

Inputs: Two.
Outputs: Two.
Effects Type(s): Gold foil reverb unit.
Delay Ranges: N/A.
Reverb/Echo Parameters: Variable, 0.8 to 5 seconds.
Operational Controls: Reverb time and lo-frequency roll-off.
Selected Standard Features: Gold foil driver, with two drivers and two pick-ups.
Frequency Response (input/output): 50 Hz to 15 kHz.
Distortion: N/A.
S/N ratio (input/output): Better than -65 dB
Pro-User Price Range: $8.815
For additional information circle #186

EVENTIDE, INC.
One Alsan Way
Little Ferry, NJ 07643
Phone: (201) 641-1200

Model SP2016

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital signal processor/reverb.
Delay Ranges: 0 to 3.2 seconds (to 1.6 seconds at full bandwidth).

Reverb/Echo Parameters: Zero to "hours" of decay time; room position; wall texture; room size; hi and low EQ controls; pre-delay; and feedback (all are program dependent).
Operational Controls: Dual input/output and dry/effect mix faders; status mode indicators; control modes; select/adjust sliders; optional user-development system (SPUD) allows user to design custom programs with external computer; optional hand-held remote control.
Selected Standard Features: 65 presets for custom setting; loop edit; flanger; multitap delay; simulated plate and room reverbs — all with adjustable parameters; alphanumeric display; and self test.
Frequency Response (input/output): 20 Hz to 16 kHz.
Distortion: THD at 1 kHz, less than 0.1%; 0.5% typical.
S/N Ratio (input/output): Dynamic range is 82 dB.
Pro-User Price Range: $6,895

Model H949 Harmonizer

Inputs: One.
Outputs: Two.
Effects Type(s): Pitch change; delay; and flanger.

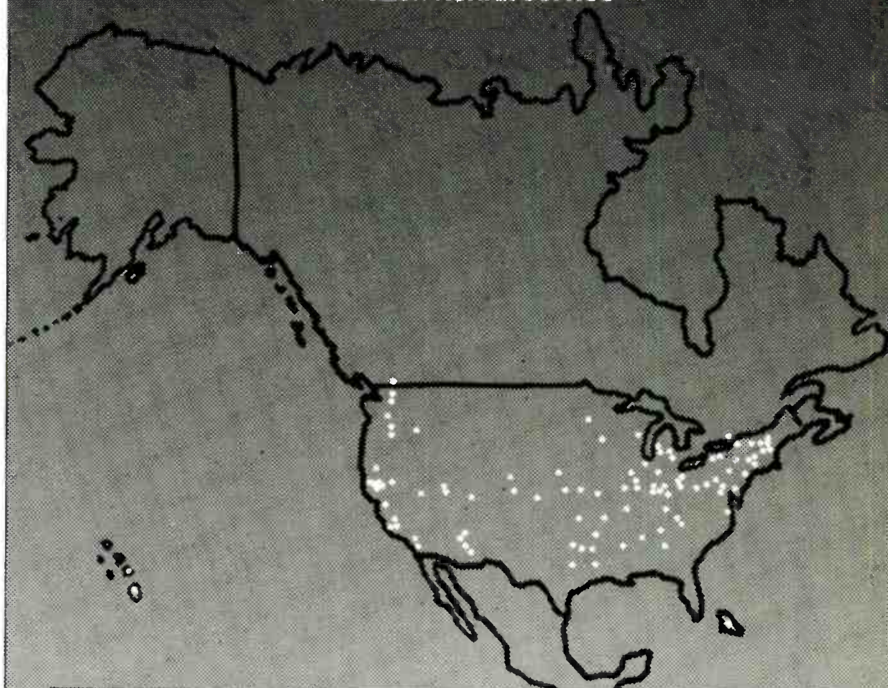
Delay Ranges: 0 to 393.75 ms in 6.25-ms step.
Reverb/Echo Parameters: Feedback and delay control.
Operational Controls: Input level control; repeat and hold button; feedback control for both main and delay outputs only; and pitch control.
Selected Standard Features: Pitch change of one octave up, or two octaves down; frequency outputs for controling tape machine motors; and time compression.
Frequency Response (input/output): 20 Hz to 15 kHz, +1 dB.
Distortion: Less than 0.15% at 1 kHz.
S/N Ratio (input/output): 96 dB dynamic range.
Pro-User Price Range: $3,500

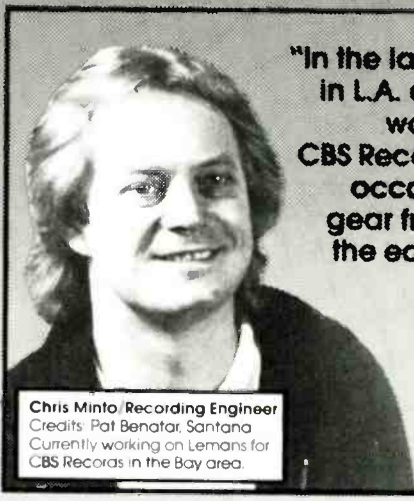
Model H910 Harmonizer

Inputs: One.
Outputs: Two.
Effects Type(s): Pitch change; delay; and echo.
Delay Ranges: 0 to 112.5 ms in 7.5-ms steps.
Reverb/Echo Parameters: Feedback and delay control.
Operational Controls: Input, feedback, anti-feedback, and pitch change controls; delay; line in/out.
Selected Standard Features: Second delay-only output (to 82.5 ms).
Frequency Response (input/output): 20 Hz to 12 kHz, +1 dB.
Distortion: Less than 0.2% at 1 kHz.
S/N Ratio (input/output): 90 dB dynamic range.
Pro-User Price Range: $1,500

Model FL201 Instant Flanger

Inputs: One.
Outputs: Two.
Effects Type(s): Flanger.
Delay Ranges: 200 microseconds to 10 ms (factory set).
Reverb/Echo Parameters: N/A.





The Directory

Operational Controls: Line in/out switch; depth; oscillator; phase shift and envelope follower.
Selected Standard Features: Remote control capability; and dual outputs for pseudo stereo effect.
Frequency Response (input/output): 50 Hz to 15 kHz, ±1 dB.
Distortion: Direct channel: less than 0.05%; delayed channel: less than 1%.
S/N Ratio (input/output): 85 dB dynamic range.
Pro-User Price Range: $700

Models JJ193/CD254

Inputs: One.
Outputs: Two and four, respectively.
Effects Type(s): Delay lines.
Delay Ranges: JJ193: 510 ms standard, optional one- or two-second versions; CD254: 254 ms. (Both units adjustable in 2-ms steps.)
Reverb/Echo Parameters: N/A.
Operational Controls: JJ193 has front-panel DIP switches; CD254 has internal switch selector.
Selected Standard Features: N/A.
Frequency Response (input/output): 50 Hz to 11 kHz, +0.5 dB.
Distortion: Less than 0.15%.
S/N Ratio (input/output): 90 dB dynamic range.
Pro-User Price Range: JJ193: $1,195 to $1,495; CD254: $895
For additional information circle #187

FOSTEX RECORDING CORP.
15431 Blackburn Avenue
Norwalk, CA 90650
Phone: (213) 921-1112

Model 3180

Inputs: Two.
Outputs: Four.
Effects Type(s): Two-channel reverb.
Delay Ranges: 24 ms pre-delay.
Reverb/Echo Parameters: RT of three seconds referenced at 1 kHz.
Operational Controls: Separate input, dry and reverb for each channel; input mix switch.
Selected Standard Features: Two channels with stereo out for each channel; input limiters; overload indicators; delay matrix provides pre-delay for six-spring system.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: N/A.
S/N Ratio (input/output): Dry: better than -80 dB unweighted; reverb: better than -58 dB unweighted.
Pro-User Price Range: $400

Model 3050

Inputs: One.
Outputs: One.
Effects Type(s): Digital delay.
Delay Ranges: 0.13 to 270 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Input level; delay range multiplier (10 steps); speed and depth modulation controls; separate dry and reverb send; output control; delay phase reverse switch.
Selected Standard Features: Eight-bit encode-decode with compander circuitry to minimize flange sweep noise; parallel foldback and auxiliary output connections; voltage control input for VC delay; limit/normal/present indicator.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: Delay THD 0.5%.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $400
For additional information circle #188

FURMAN SOUND, INC.
30 Rich Street
Greenbrea, CA 94904
Phone: (415) 927-1225

Model RV-1

Inputs: One.
Outputs: One.
Effects Type(s): Spring reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: Pre-delay ranges of 33, 37, and 41 ms.

Operational Controls: Semi-parametric mid-EQ; shelving treble EQ; two output level controls (wet/dry); ground lift; and input level control.
Selected Standard Features: Input limiter; low- and high-level inputs; and footswitch jack.
Frequency Response (input/output): Direct: 20 Hz to 20 kHz, -0.5/+0 dB; reverb: 45 Hz to 7 kHz, with closely spaced peaks and dips.
Distortion: Direct: less than 0.01%.
S/N Ratio (input/output): Direct: better than -109 dB; reverb: -85 dB.
Pro-User Price Range: Mono: $321; Stereo: $535
For additional information circle #189

IBANEZ
U.S. Distributors: Hoshino USA
and Chesebro Music Co.
1716 Winchester Road
Bensalem, PA 19020
Phone: (215) 638-8670

DM1100 Digital Delay

Inputs: Two.
Outputs: Two.
Effects Type(s): Time delay effects (modulated and unmodulated).

Delay Ranges: 7 ms to 3,600 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Input level; delay tone; delay time; modulation width and speed; feedback; dry and delay levels.
Selected Standard Features: Mike/line level switching; normal and inverted outputs; and repeat-hold capabilities.
Frequency Response (input/output): Delay: 30 Hz to 8 kHz, -3/+0.5 dB.
Distortion: THD less than 1% on all ranges.
S/N Ratio (input/output): EIN: -95 dB (IHF-A).
Pro-User Price Range: $329

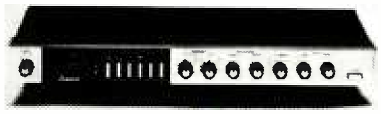
DM2000 Digital Delay

Inputs: Two.
Outputs: Four.
Effects Type(s): Time delay effects (modulated and unmodulated).
Delay Ranges: 0.1 ms to 2,047 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Input level; delay EQ; delay time; modulation speed and width; feedback; dry and delay levels; and preset select.
Selected Standard Features: Eight presets; remote switching; normal and inverted outputs; repeat-hold capabilities.
Frequency Response (input/output): Delay: 10 Hz to 20 kHz, -3/+0.5 dB.
Distortion: THD less than 0.2% on all ranges.
S/N Ratio (input/output): EIN: -95 dBv (IHF-A).
Pro-User Price Range: $449

HD1500 Harmonics/Delay

Inputs: Two.
Outputs: Three.
Effects Type(s): Pitch shift and time delay.
Delay Ranges: 0 to 504 ms; shift range: +1300 to -1300 cents.

Reverb/Echo Parameters: N/A.
Operational Controls: Input level; delay EQ; delay time; modulation speed and width; feedback; dry and delay levels.



Selected Standard Features: Mike/line level switching; dry and mix outputs; three preset shifts; with optional remote footswitch.
Frequency Response (input/output): Delay: 30 Hz to 8 kHz, referenced at 2 and 4 kHz; harmonics: 30 Hz to 8 kHz.
Distortion: THD less than 1% on all ranges.
S/N Ratio (input/output): EQ input noise: -93 dBv (IHF-A).
Pro-User Price Range: $559
For additional information circle #190

INDUSTRIAL RESEARCH PRODUCTS, INC.
321 Bond Street
Elk Grove Village, IL 60007
Phone: (312) 439-3600

Model DF-4015

Inputs: One.
Outputs: Up to four.
Effects Type(s): Digital delay.
Delay Ranges: Zero to 192 ms, in 3-ms increments.
Reverb/Echo Parameters: N/A.
Operational Controls: Input level; and thumbwheel switches.
Selected Standard Features: Red and green clipping LEDs; transformer balanced in and out; security panel.
Frequency Response (input/output): 12 Hz to 15 kHz, ±2 dB.
Distortion: Less that 0.012% at 1 kHz, +4 dBm.
S/N Ratio (input/output): Better than -90 dB unweighted.
Pro-User Price Range: $1,500 to $3,000
For additional information circle #191

KLARK-TEKNIK, INC.
262a Eastern Parkway
Farmingdale, NY 11735
Phone: (516) 249-3660

Models DN700/701

Inputs: One.
Outputs: Three.
Effects Type(s): N/A.
Delay Ranges: 0 to 434 ms; and 0 to 1.7 seconds, respectively.
Reverb/Echo Parameters: N/A.
Operational Controls: Input level; independent delay control per channel.
Selected Standard Features: Electronic safe switch; LED readout; and headroom indicator.
Frequency Response (input/output): 20 Hz to 15 kHz.
Distortion: 0.01%.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: 700: $1,295; 701: $1,795

Models DN780

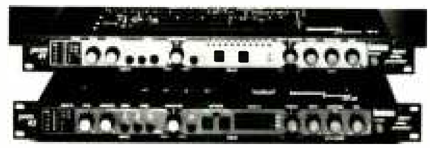
Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb-hall; plate; chamber; room; and gated.
Delay Ranges: Pre-delay 0 to 99 ms; RT zero to 99 seconds.
Reverb/Echo Parameters: Pre-delay; early reflections-pattern; and level.
Operational Controls: HF and LF equalization;

decay; room size; input; and reverb time.
Selected Standard Features: Remote control; 16 program sequencing; 50 user memories; and five special effects.
Frequency Response (input/output): 20 Hz to 12 kHz.
Distortion: 0.03%.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: $5,500
For additional information circle #192

LEXICON, INC
60 Turner Street
Waltham, MA 02154
Phone: (617) 891-6790

Model PCM-41

Inputs: One.
Outputs: One.
Effects Type(s): Digital delay.
Delay Ranges: Delay #1 range: normal delay of 1 to 200 ms at full bandwidth, double delay 400 ms at 6k Hz; delay #2 range: normal delay of 400 ms full bandwidth, double delay 800 ms at 6 kHz.



Operational Controls: Input; feedback; output mix; delay multiply; VCO depth, waveform, and rate; delay select and range; feedback filters and invert; infinite repeat.
Frequency Response (input/output): 20 Hz to 15 kHz, +0/-1 dB in #1 delay mode.
Distortion: 0.1% typical over bandpass 20 Hz to 15 kHz.
S/N Ratio (input/output): Dynamic range better than 90 dB, 20 Hz to 20 kHz.
Pro-User Price Range: $715

Model PCM-42

Inputs: One.
Outputs: One.
Effects Type(s): Digital delay line.
Delay Ranges: Standard memory 16 kHz mode: 0.5 is 400 ms and 1.5 is 1.2 seconds; 6 kHz mode: 0.5 is 800 ms and 1.5 is 2.4 seconds.

For additional information circle #87

The Directory

Reverb/Echo Parameters: N/A.
Operational Controls: Input; feedback; high cut; output mix; modulation; VCO sweep, depth, waveform, and rate; clock indicator; 6 kHz bandwidth indicator; and infinite repeat.
Selected Standard Features: Slap echo; ambient echo; long echo; doubling; flanging; resonance; vibrato; pitch shifting; and clock output.
Frequency Response (input/output): 20 Hz to 16 kHz, +0.5/-3 dB in first delay mode.
Distortion: 0.2% typical over bandpass 20 Hz to 15 kHz.
S/N Ratio (input/output): Dynamic range better than 90 dB, 20 Hz to 20 kHz.
Pro-User Price Range: 2.4 second delay: $1,000; 4.8 second delay: $1,235

Model 95 Prime Time II

Inputs: One (plus one auxiliary).
Outputs: Two.
Effects Type(s): Digital delay processor.
Delay Ranges: Standard memory 960 ms; double 1.92 seconds; full memory option in normal mode 3.8 seconds; in double mode 7.68 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Two separate dry controls; delay modulation depth, shape, and rate; infinite repeat; dynamic recirculation; input level; A and B feedback; feedback filter output mix; master output.
Selected Standard Features: Dynamic recirculation; doubling; tripling; chorus; flanging; resonance; clock output; slap echo; long echo.
Frequency Response (input/output): 20 Hz to 16 kHz, +0.5/-2 dB.
Distortion: 0.1% maximum, 20 Hz to 10 kHz.
S/N Ratio (input/output): Better than -90 typical, 20 Hz to 20 kHz.
Pro-User Price Range: $1,980

Model 97 Super Prime Time

Inputs: One (plus one auxiliary).
Outputs: Two.
Effects Type(s): Digital delay processor.
Delay Ranges: Standard memory: 0.2 to 640 ms.
Reverb/Echo Parameters: N/A.

Operational Controls: Delay modulation depth, shape, and rate; main and auxiliary levels; delay output mix; recirculation controls with filter.
Selected Standard Features: Flanging; resonant flanging; doubling; tripling; chorus; slap echo; short echo; long echo; and 32 user registers.
Frequency Response (input/output): 20 Hz to 20 kHz, +1/-2 dB.
Distortion: 0.05% maximum at 1 kHz.
S/N Ratio (input/output): Better than -85 dB, 20 Hz to 20 kHz.
Pro-User Price Range: $3,170

Model PCM-60

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb.
Delay Ranges: Room program 0.3 to 3.8 seconds; plate program 0.2 to 4.5 seconds.
Reverb/Echo Parameters: Two programs; four sizes; four reverb times; and frequency contours.
Operational Controls: Input level; mix control; output control; input and output level range select; and bypass.
Selected Standard Features: Plate/room programs with various sizes and reverb times.
Frequency Response (input/output): 20 Hz to 10 kHz, ±1 dB.
Distortion: Better than or equal to 0.05%, maximum at 1 kHz.
S/N Ratio (input/output): Better than -80 dB, 20 Hz to 20 kHz.
Pro-User Price Range: $1,500

Model 200

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital reverb.
Delay Ranges: 0.1 to 99 seconds, plus infinite reverb.
Reverb/Echo Parameters: Room size; reverb time; pre delay; and frequency contour.
Operational Controls: Input levels; output mix; input and output mute.
Selected Standard Features: Six standard programs with 54 variations, plus 10 user storage registers.
Frequency Response (input/output): 20 Hz to 10 kHz, ±1 dB.
Distortion: 0.07%, maximum at 1 kHz.
S/N Ratio (input/output): Better than -84 dB, 20 Hz to 20 kHz.
Pro-User Price Range: $4,800

Model 224XL

Inputs: Two.
Outputs: Four.
Effects Type(s): Digital reverb/effects processor.
Delay Ranges: 0.1 ms to 1.96 seconds effects program; 0.1 to 70 seconds plus infinite reverb on reverb programs.

Reverb/Echo Parameters: Over 32 user adjustable parameters.
Operational Controls: LARC (Lexicon Alphanumeric Remote Control), fully programmable.
Selected Standard Features: Alphanumeric listing more than 60 reverb and effects variations.
Frequency Response (input/output): 20 Hz to 15 kHz, +1.5 dB.
Distortion: 0.07%, maximum for all reverb times between 0 and 35 seconds.
S/N Ratio (input/output): Better than -84 dB, 20 Hz to 20 kHz for all reverb times between 0 and 10 seconds.
Pro-User Price Range: $12,500
For additional information circle #193

MARSHALL ELECTRONIC
P.O. Box 438
Brooklandville, MD 21022
Phone: (301) 484-2220

Model AES-357

Inputs: Two.
Outputs: Four.
Effects Type(s): Ambience effects system: room simulation, stereo synthesis, and reverb processing.
Delay Ranges: 35 microseconds to "several seconds."
Reverb/Echo Parameters: 3,000 program locations; 132 output taps; up to 1,200 initial reflections.
Operational Controls: Room size; room shape; ambience; stereo spread; width; depth; echo density; "image type."
Selected Standard Features: 100 types of stereo image generation with 1,000 programs; 100 room-shape size combinations of room simulation and ambience generation; 100 post-reverb processor images; RS232 interface.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -95 dB.
Pro-User Price Range: $2,995

AR-300 Tape Eliminator

Inputs: One.
Outputs: One.
Effects Type(s): "Replicates the sound of a quality tape recorder without the noise."
Delay Ranges: Interhead delay 20 to 75 ms at 30 ips; 40 to 150 at 15 ips; and 80 to 300 ms at 7.5 ips.
Reverb/Echo Parameters: Three tape speeds with VSO; internal adjustment of HF EQ curve and bias; choice (blend) of even or odd harmonic dominance at tape saturation for each tape speed.
Operational Controls: Tape speed and VSO.
Selected Standard Features: Tape slap for "golden ears" effect.
Frequency Response (input/output): 15 Hz to 15 kHz.
Distortion: As per over-bias and flux settings.
S/N Ratio (input/output): Better than -102 dB.
Pro-User Price Range: $1,295

5402 Time Modulator

Inputs: One.
Outputs: One
One.
Effects Type(s): Positive/negative flange; ADT; vibrato; pitch detune; echo; delay; vocoder effects; resonant drum tuning; plus combinations of above.
Delay Ranges: 150 microseconds to 400 ms.
Reverb/Echo Parameters: Six delay ranges in combination with mix-sweepable time, with no quantization steps.
Operational Controls: Dry feed; three positive and negative output taps; level control; feedback; time, depth, and speed modulation; waveform.
Selected Standard Features: Modulation input; full multi-tap output mixing; 2000 to 1 delay range sweep continue; total analog signal path for no digital distortion.
Frequency Response (input/output): Bandwidth 20 Hz to 15 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $1,995
For additional information circle #194

NEPTUNE ELECTRONICS, INC.
934 N.E. 25th Street
Portland, OR 97232
Phone: (503) 232-4445

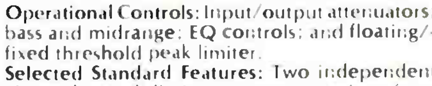
Model 351

Inputs: One.
Outputs: One.
Effects Type(s): Spring reverb.
Delay Ranges: Variable.
Reverb/Echo Parameters: N/A.
Operational Controls: Input gain; output level; five-band graphic EQ; in/out switching; mute; and signal activated reverb.
Selected Standard Features: Signal activated reverb helps eliminate howling and spring slap due to external vibrations.
Frequency Response (input/output): Direct: 20 Hz to 14 kHz.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $375
For additional information circle #195

ORBAN ASSOCIATES, INC.
645 Bryant Street
San Francisco, CA 94107
Phone: (415) 957-1067

Model 111B

Inputs: Two.
Outputs: Two.
Effects Type(s): Spring reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.



Operational Controls: Input/output attenuators; bass and midrange; EQ controls; and floating/fixed threshold peak limiter.
Selected Standard Features: Two independent channels; peak limiter to protect springs from transients; EQ controls to tailor frequency response.
Frequency Response (input/output): 25 Hz to 16 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -76 dB.
Pro-User Price Range: $899
For additional information circle #196

PHOENIX AUDIO LAB, INC.
91 Elm Street
Manchester, CT 06040
Phone: (203) 649-1199

Loft Model 450

Inputs: One.
Outputs: Two.
Effects Type(s): Delay/flanger.
Delay Ranges: 160 to 360 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: Reverb input, amount, output; regeneration (EQ shift); and LED indicators.
Selected Standard Features: Muscial instrument pre-amp; rear-panel foot pedal jack; jack and XLR connectors on inputs and outputs; rack mountable.
Frequency Response (input/output): N/A.
Distortion: Flange is 0.2%; delay 0.8%.
S/N Ratio (input/output): Better than -80 dB, A-weighted.
Pro-User Price Range: $399 to $549.
For additional information circle #198

PUBLISON AMERICA, INC.
6464 Sunset Blvd. Suite #980
Hollywood, CA 90028
Phone: (213) 460-6355

Internal Machine 90

Inputs: Two.
Outputs: Four.
Effects Type(s): Reverberation; pitch change; and time compression on two separate channels.
Delay Ranges: 0.04 ms to 200 seconds.

Reverb/Echo Parameters: Volume; bass; dual treble coefficients; three primary reflections in delay and gain.
Operational Controls: Controlled by digital keyboard.
Selected Standard Features: Two entirely independent stereo reverber channels; two pitch changes; two 20-second memory; 200 memories for parameters; MIDI interface.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: THD less than 0.03%.
S/N Ratio (input/output): Dynamic range: 96 dB.
Pro-User Price Range: $10,900
For additional information circle #199

QUANTEC U.S.
U.S. Distributor:
Europa Technologies
1638 West Washington Blvd.
Venice, CA 90291
Phone: (213) 392-4985

Models QRS & QRS/L

Inputs: Two and one, respectively.
Outputs: Four and two, respectively.
Effects Type(s): Digital Room Simulator.
Delay Ranges: 0.1 to 100 seconds (up to 400 seconds at 40 Hz).
Reverb/Echo Parameters: Room size; 1 separate HF and LF RT 60 times; to one million cubic meters.
Operational Controls: Room size; level and delay for first reflection; and enhance and freeze program (loop).
Selected Standard Features: LED bargraph; XLR connectors; infinite rotary control pot; 64 programmable memories; optional infrared remote controller and JL Cooper MIDI interface.
Frequency Response (input/output): 20 Hz to 8 kHz, +1/-3 dB.
Distortion: N/A.
S/N Ratio (input/output): Better than -88 dB.

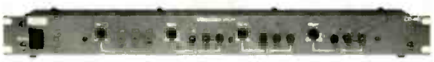
The Directory

Pro-User Price Range: QRS: $7,995; QRS/L: $4,495
For additional information circle #200

RANE CORP.
6510 216th SW Street
Mountlake Terrace, WA 98043
Phone: (206) 774-7309

Model CD 48

Inputs: Four.
Outputs: Four.
Effects Type(s): Analog delay.
Delay Ranges: 0 to 8 ms, continuously variable.



Reverb/Echo Parameters: N/A.
Operational Controls: Each channel has a 0 to 2 ms control and three pushbuttons for +2 ms each.
Selected Standard Features: For phase alignment of multiway speaker system.
Frequency Response (input/output): 20 Hz to 20 kHz, ±1 dB.
Distortion: Less than 0.09% at +4 dBm.
S/N Ratio (input/output): 106 dB at +20 dBm.
Pro-User Price Range: $349
For additional information circle #201

ROLAND CORP.
7200 Dominion Circle
Los Angeles, CA 90040
Phone: (213) 685-5141

SDE2500

Inputs: Three.
Outputs: Eight.
Effects Type(s): MIDI programmable digital delay.
Delay Ranges: 0 to 750 ms.
Reverb/Echo Parameters: 15 bit A/D/A converter.
Operational Controls: Input gain; delay; modulation (rate/depth); memory; delay time; write-copy; program channel.
Selected Standard Features: 64 programs; 128 MIDI positions.
Frequency Response (input/output): 10 Hz to 17 kHz.
Distortion: Direct: 0.008%; delay: 0.05%.
S/N Ratio (input/output): Better than -96 dB.
Pro-User Price Range: $795

SRV2000

Inputs: Two.
Outputs: Three.
Effects Type(s): MIDI programmable digital reverb.
Delay Ranges: Reverb time: 0.1 to 99.9 seconds; pre-delay 0 to 300 ms.
Reverb/Echo Parameters: First initial response (FIR); pre-delay.
Operational Controls: Input gain; MIDI; memory number; EQ; manual EQ; pre-delay; reverb time; HF damp; copy/write; room size.
Selected Standard Features: FIR; 32 programs; MIDI channel selectable; and MIDI in/thru.
Frequency Response (input/output): N/A.
Distortion: N/A.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $1,495

SDE3000

Inputs: Four.
Outputs: Eight.
Effects Type(s): MIDI programmable digital delay.
Delay Ranges: 0 ms to 4.5 seconds.
Reverb/Echo Parameters: All front-panel controls programmable.
Operational Controls: Input gain; MIDI; memory ×8; filter; ×2; phase; modulation; feedback phase; delay output; rate; depth; time.
Selected Standard Features: Feedback effect loop; "playmate"; modulation CV in; remote delay switching.
Frequency Response (input/output): 10 Hz to 17 kHz.
Distortion: 0.008%.
S/N Ratio (input/output): Better than -88 dB.
Pro-User Price Range: $1,095

SDE1000

Inputs: Two.
Outputs: Six.
Effects Type(s): Digital delay.
Delay Ranges: 0 ms to 1.125 seconds.
Reverb/Echo Parameters: Four programs.
Operational Controls: Input gain; feedback; delay out; modulation; time delay phase; memory.
Selected Standard Features: User programmable; "playmate" and hold features; modulation (cv) input.
Frequency Response (input/output): 10 Hz to 17 kHz.
Distortion: 0.08%.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $499

SPE1000

Inputs: Three.
Outputs: Seven.
Effects Type(s): Digital delay.
Delay Ranges: 0 to 640 ms.
Reverb/Echo Parameters: 0 to 10 ms in 0.10-ms steps; above 10 ms in 1 ms steps.
Operational Controls: Input gain; feedback; single repeat switch; delay out; phase invert; delay ×2; and modulation.
Selected Standard Features: Rack mount; +12dB input gain; features for doubling delay time.
Frequency Response (input/output): 10 Hz to 16 kHz.
Distortion: 0.05%.
S/N Ratio (input/output): Better than -112 dB.
Pro-User Price Range: $1,295
For additional information circle #202

SCV, INC.
414 North Sparks Street
Burbank, CA 91506
Phone: (818) 843-7567

Model RBS2

Inputs: Two.
Outputs: Two.
Effects Type(s): Spring reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: Delay time: 3.5 seconds; first reflection: 30 ms.

Operational Controls: Reverb input, amount, output; spin (regeneration); low filter; bass; treble; and single/stereo input switch.
Selected Standard Features: Six spring channels; left and right drive level VU LEDs; two-piece package (electronics, plus spring reverb).
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: THD less than 0.1%; IMD 0.01%.
S/N Ratio (input/output): Better than -70 dB.
Pro-User Price Range: $925
For additional information circle #203

SONY CORP.
One Sony Drive
Park Ridge, NJ 07656
Phone: (201) 930-1000

DRE2000A Digital Reverb

Inputs: Two.
Outputs: Two.
Effects Type(s): Modes: four reverb; two echo; and two delay.
Delay Ranges: Mode 1: channel one and two has 0 to 999 ms in 1-ms steps; mode 2: channel one has 0 to 999 ms in 1 ms steps; channel two has 0 to 499 ms in 1-ms steps.
Reverb/Echo Parameters: REVERB: 0 to 9.9 seconds, in 0.1-second steps; early reflection level of 0.1 to 0.9 in 0.1-steps; pre-delay 0 to 255 ms in 1-ms steps; sub-reverb level 0.1 to 0.9 in 0.1-steps, delay 0 to 255 ms in 1-ms steps; ECHO: echo-1 feedback factor 0 to 1.9 in 0.1-steps for channels 1 and 2; echo-2 0 to 999 ms in 1-ms steps for channel one, and 0 to 499 ms to 1-ms steps for channel two.
Operational Controls: Touch keys; direct access with numerical keypad; LED displays; and headroom indicator.
Selected Standard Features: Non-volatile user memory for 10 programs; one parameter memory for each master program; and eight factory presets.
Frequency Response (input/output): 30 Hz to 13 kHz, -3 +1 dB.
Distortion: Less than 0.3% THD at 1 kHz.
S/N Ratio (input/output): Better than -78 dB on reverb and echo; -81 dB for delay.
Pro-User Price Range: $9,500
For additional information circle #204

STUDIO TECHNOLOGIES
7250 N. Cicero Avenue
Lincolnwood, IL 60646
Phone: (312) 676-9177

Ecoplate III

Inputs: Two.
Outputs: Two.
Effects Type(s): Plate reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: RT 60 from 0.5 to 5 seconds.
Operational Controls: High and low EQ.
Selected Standard Features: N/A.
Frequency Response (input/output): 80 Hz to 20 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -65 dB.
Pro-User Price Range: $1,695

AN-2 StereoSimulator

Inputs: Two.
Outputs: Two.
Effects Type(s): Stereo synthesizer.
Delay Ranges: N/A.
Reverb/Echo Parameters: N/A.
Operational Controls: Stereo width and depth; gain; and modulation.
Selected Standard Features: N/A.
Frequency Response (input/output): 20 Hz to 15 kHz, ±2 dB.
Distortion: 0.2% THD.
S/N Ratio (input/output): Better than -70 dB.
Pro-User Price Range: $650
For additional information circle #205

SYCHRONOUS TECHNOLOGIES
1020 West Wilshire Blvd.
Oklahoma City, OK 73116
Phone: (405) 843-9626

Hot Springs

Inputs: One.
Outputs: One.
Effects Type(s): Springs reverb with equalizer.
Delay Ranges: N/A.

Reverb/Echo Parameters: Frequency select; resonance; and boost/cut.
Operational Controls: N/A.
Selected Standard Features: Paramteric EQ and overload EQ.
Frequency Response (input/output): 20 Hz to 8 kHz.

The Directory

Distortion: N/A.
S/N Ratio (input/output): Better than -66 dB.
Pro-User Price Range: $499
For additional information circle #206

UNICORD
89 Frost Street
Westbury, NY 11590
Phone: (516) 333-9100

Korg SDD1000

Inputs: One.
Outputs: Three.
Effects Type(s): Digital delay.
Delay Ranges: 1 ms to 2.048 seconds.
Reverb/Echo Parameters: Delay time; frequency and intensity modulation; effect level; feedback; high-cut feedback filters.
Operational Controls: Input level; direct output level; effect level; delay time; factor; feedback; hi-cut; and frequency input attenuator with unity gain control.
Selected Standard Features: Sampling up to 2.048 seconds; infinite hold; external setting of delay time.
Frequency Response (input/output): 30 Hz to 10 kHz at 1.024 seconds; 30 Hz to 5 kHz at 2.048 seconds.
Distortion: 0.05% direct; 0.1% effect.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $395

Korg GR1

Inputs: Two.
Outputs: Two.
Effects Type(s): Spring reverb.
Delay Ranges: N/A.
Reverb/Echo Parameters: Gate threshold and decay time; three-band EQ.
Operational Controls: Input level; direct output level; gate on/off switch; threshold; decay time; LF, MF, HF EQ; direct output level; reverb level; incremental control for all delay parameters; input attenuator with unity gain control.
Selected Standard Features: Integral gate section allows setting of decay time; EQ section.
Frequency Response (input/output): 200 Hz to 4.5 kHz.
Distortion: N/A.
S/N Ratio (input/output): Better than -60 dB for effect; -8 for direct.
Pro-User Price Range: $295

Korg SDD2000

Inputs: One.
Outputs: Three.
Effects Type(s): MIDI programmable digital delay.
Delay Ranges: 0 ms to 4.368 seconds.
Reverb/Echo Parameters: Delay time; frequency and intensity modulation; effect level; feedback.
Operational Controls: Input level; direct output level; incremental control for all delay parameters; input attenuator with unity gain control.
Selected Standard Features: Sampling up to 4.368 seconds; MIDI program change; 64 programmable settings; MIDI playback of sample.
Frequency Response (input/output): 30 Hz to 18 kHz at 1.092 seconds; 30 Hz to 4.5 kHz at 4.368 seconds.
Distortion: 0.05% direct; 0.1% effect.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $695
For additional information circle #208

URSA MAJOR, INC.
P.O. Box 28
Boston, MA 02258
Phone: (617) 924-7697

Space Station SST-282

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb/effects.
Delay Ranges: Echo delay time: 0 to 256 ms.
Reverb/Echo Parameters: 0 to 3.5 seconds of delay time.
Operational Controls: Knobs and push button controls throughout; audition delay mixer for taps 1 thru 8; pushbutton control for room selection one thru four; comb filters; pre-set programs.
Selected Standard Features: N/A.
Frequency Response (input/output): 20 Hz to 7 kHz.
Distortion: Total distortion and noise 0.1%, typical.
S/N Ratio (input/output): Better than -80 dB.
Pro-User Price Range: $2,195

Stargate 626/323

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital reverb/effects.
Delay Ranges: Pre-delay: 0 to 320 ms.
Reverb/Echo Parameters: 323: 0 to 10 second decay time in eight room programs; 626: 0 to 20 second decay in eight effects programs.
Operational Controls: LF decay, HF decay; direct and reverb mix controls; room selector; pre-delay control; input mode; dry only; reverb clear.
Selected Standard Features: LED level and numeric displays.
Frequency Response (input/output): 15 kHz for all rooms at all decay times; sampling rate is 32 kHz.
Distortion: Total distortion and noise 0.1%, typical.
S/N Ratio (input/output): 80 dB dynamic range.
Pro-User Price Range: 626: $2,500; 623: $2,000

Model MSP-126

Inputs: Two.
Outputs: Two.
Effects Type(s): Multi-tap stereo processor; comb-stereo processing; room early reflections; delay clusters; pan mode; digital delay; repeats; and musical scale in steps.
Delay Ranges: 100 to 360 ms.
Reverb/Echo Parameters: N/A.
Operational Controls: 16-character alphanu-

meric display shows knobbed control of parameter one: 16 delay variations of each mode parameter two: 16 amplitude variations.
Selected Standard Features: LED level display and bypass control.
Frequency Response (input/output): Bandwidth 20 kHz, 44.1 kHz sampling rate.
Distortion: Total distortion and noise 0.1%, typical.
S/N Ratio (input/output): 80 dB dynamic range.
Pro-User Price Range: $2,000
For additional information circle #209

YAMAHA INTERNATIONAL CORP.
6600 Orangethorpe
Buena Park, CA 90620
Phone: (714) 522-9011

Model D1500

Inputs: One.
Outputs: Two.
Effects Type(s): Digital delay.
Delay Ranges: 0 to 1.023 seconds.
Reverb/Echo Parameters: Delay time and LFO.

Operational Controls: Delay time lowpass filter: level: signal invert: LFO rate: wave and depth mix: recall memorized for each preset.
Selected Standard Features: 16 memories: MIDI recall of presets: bypass and repeat hold by front panel or footswitch: jack and XLR connection: input and output levels.
Frequency Response (input/output): Input: 20 Hz to 20 kHz: delay: 20 Hz to 18 kHz.
Distortion: Input: 0.008%; delay 0.08%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $895

Model REV-1

Inputs: One.
Outputs: Two.
Effects Type(s): Digital reverb
Delay Ranges: Early reflections: 0 to 255 ms: subsequent reverb: 0 to 600 ms.

Reverb/Echo Parameters: Up to 40 early reflections, and up to 99.9 seconds of reverb.
Operational Controls: Low and high pass filters: early reflection (mode, number of early reflections, and room liveness): reverb (mode, level in four frequency ranges).
Selected Standard Features: 99 total memories, with 30 pre-programmed, 60 user-programmable, and nine in remote controller: edit-recall function: RCR-1 remote control with LCD readout and eight screen menus.
Frequency Response (input/output): Input: 20 Hz to 18 kHz.
Distortion: Less than 0.3%.
S/N Ratio (input/output): Better than -85 dB.
Pro-User Price Range: $11,900

Model REV-7

Inputs: Two.
Outputs: Two.
Effects Type(s): Digital reverb.
Delay Ranges: Up to 100 ms: up to 10 seconds of reverberation.
Reverb/Echo Parameters: Early reflections: subsequent reverberation: pre-early reflection: and pre-reverb delays.
Operational Controls: N/A.
Selected Standard Features: Mono/stereo modes: three-band semi-parametric EQ; MIDI recall of presets: 10 presets: 60 user memories: and remote control.
Frequency Response (input/output): Input: 20 Hz to 20 kHz.

Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -66 dBm.
Pro-User Price Range: N/A

Model R1000

Inputs: One.
Outputs: One.
Effects Type(s): Digital reverberator.
Delay Ranges: Reverb time: 1.5, 1.6, 2.3, and 2.4 seconds.
Reverb/Echo Parameters: N/A.
Operational Controls: Input: output: mix controls: and bypass.
Selected Standard Features: Semi-parameteric EQ with bypass, -20 or +4 dB output.
Frequency Response (input/output): Input: 20 Hz to 20 kHz.
Distortion: N/A.
Seconds/N Ratio (input/output): Better than -60 dB.
Pro-User Price Range: $795

Model YDD-2600

Inputs: Four.
Outputs: Eight.
Effects Type(s): Digital delay.
Delay Ranges: 0 to 2.660 seconds (one input): 0 to 1.333 seconds (two inputs): and 0 to 655 ms (4 inputs).
Reverb/Echo Parameters: N/A.
Operational Controls: Level: memory: delay time entered in ms or distance.
Standard Features: Various in/out configurations available: remote control: 12 memories: LCD display.
Frequency Response (input/output): 20 Hz to 20 kHz.
Distortion: Less than 0.03%.
S/N Ratio (input/output): Better than -90 dB.
Pro-User Price Range: $7,700
For additional information circle #207

SONY APR-5002 ANALOG TWO-TRACK RECORDER

Reviewed by Peter Butt

The APR-5002 is the first new product to emerge from the former MCI Fort-Lauderdale facility since that firm was sold to Sony more than two years ago. At only 91 pounds, the APR-5002 is smaller and lighter than other professional quality mastering recorders have ever been. The machine accommodates reels of up to 12½ inches diameter, and quarter- or half-inch tape widths. The APR-5000 is supplied in a number of configurations, from monophonic, full-track ¼-inch, to two-track NAB-or DIN-track geometry, to two-track ½-inch format in two speed configurations, covering the range from 3.75 to 30 ips. Equalization characteristics are IEC, NAB, and AES selectable. The model tested is the one featuring the higher three common analog tape speeds: 7.5, 15 and 30 ips with NAB geometry.

The machine is small enough to be used as a table-top unit or as console mounted in the optional SU-14 stand, allowing deck orientation to be horizontal or 15-degree tilt.

A tertiary central tape track intended for timecode purposes is to be available on APR-5003 two-track models within a few months. The exact nature of this feature remains obscure, although it is to include an internal SMPTE generator and the capability for "chase" mode operation. No mention is made of timecode offset accommodation capability, an important feature for real-life synchronization applications. The preliminary service manual accompanying the machine refers to a "special" timecode track equalization used with record/play functions provided by a fourth head situated between the erase and record heads. No further description of the timecode recording process is given, although one would infer that it is based on linear analog recording reproduction, rather than a modulated carrier to permit data scanning at high tape shuttle speeds. The machine is configured to facilitate synchronized applications with all deck and audio data channel status control functions accessible through parallel and RS-422 serial ports at the rear of the deck assembly. Flux-frequency references for timecode track alignment have not been indicated thus far.

The APR-5000 is microprocessor controlled; evidently, the CPU is time-shared between deck-control duties and signal-channel functions. All signal-channel functions, except manually uncalibrated input and output gain adjustments, are controlled by the CPU. All audio control settings are indicated in a two-digit hexadecimal code, giving 256 discrete values for input-monitor level, record level, record-HF boost, record-bias drive, reproduce and sync gain, LF

Figure 1: Input impedance magnitude versus frequency for input port 1, typical.

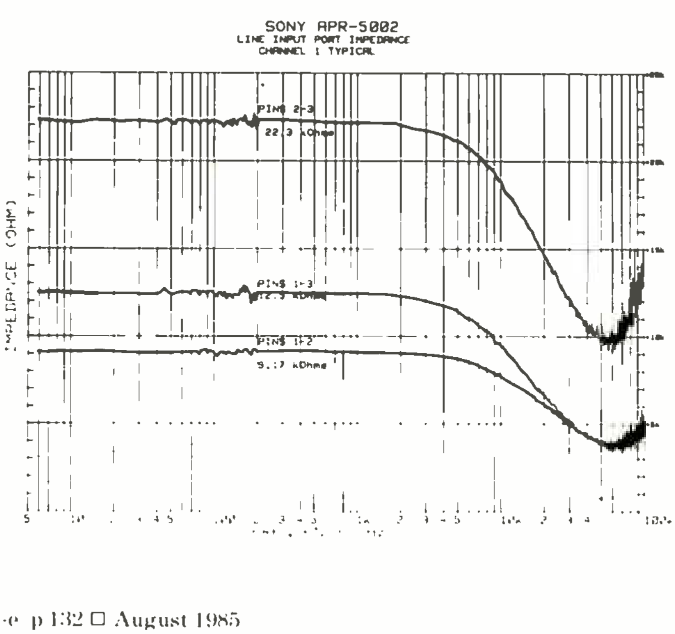
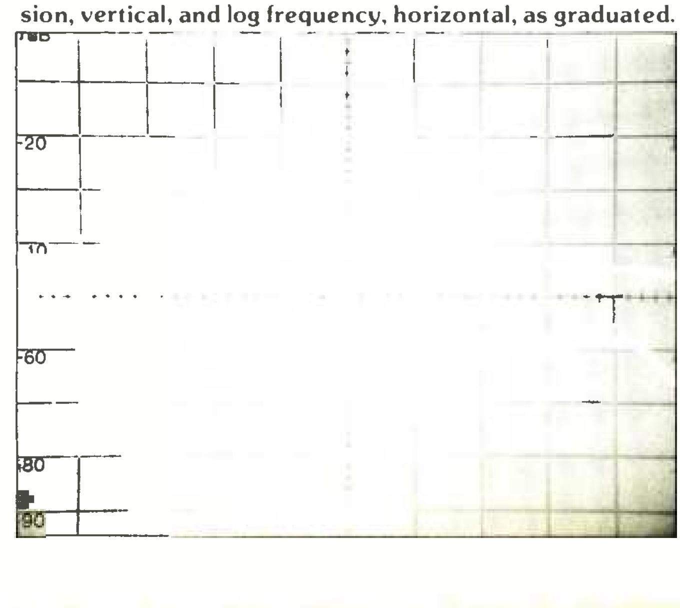


Figure 2: Input port common-mode response. Top trace is the normal-mode response of channels 1 and 2 overlaid; top graticule reference is +4 dBv. Common mode responses for channels 1 and 2 are shown as the center and bottom traces, respectively. Scale factors are 10 dB/division, vertical, and log frequency, horizontal, as graduated.



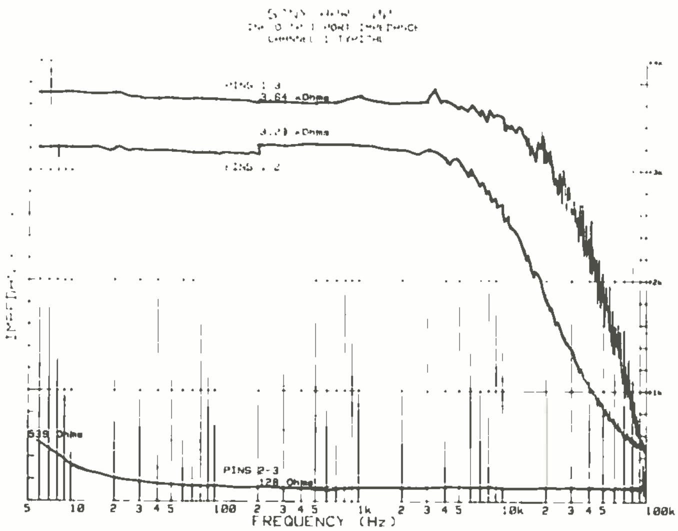


Figure 3: Output impedance magnitude versus frequency for output port 1, typical.

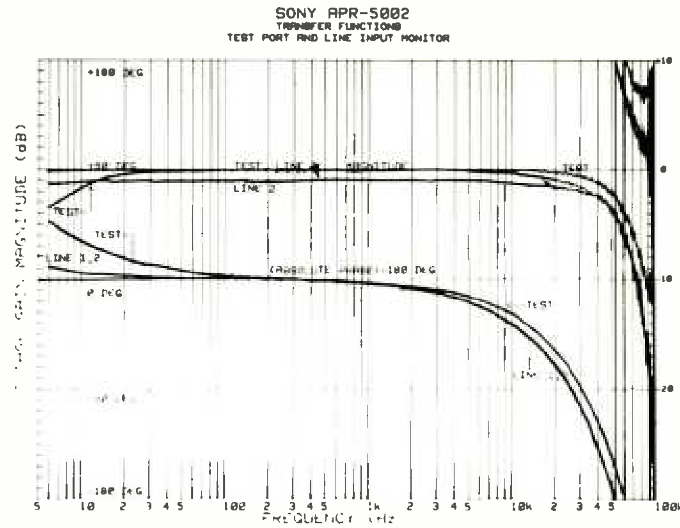


Figure 4: Input monitor transfer functions for main audio channels and Calibration Input/Output ports. Main audio channels phase contours are rotated -180 degrees for clarity.

and HF equalization. (Oddly, the hex code for maximum setting values is "00," while minimum values are indicated by "FF." The numerical weight of the indication is inversely related to the magnitude of the indicated variable.)

The unit's service manual gives the use of each of the control and alignment functions in a step-by-step sequence, where proper CPU responses are indicated, as well as the precise stimulus sequences required. The sequential and conditional precision demanded by semi-smart devices is difficult to convey by general discussion. The hand-holding instructional approach taken by the writers of the APR-5000 manual is appreciated, and should be emulated by other manufacturers as well. All visual annunciators or indicators are either LEDs or electroluminescent devices that need no external lighting for functional visibility, and can therefore be clearly seen in low ambient lighting. LCDs don't serve well in the dark.

Provision is made for the retention of data for up to three audio electronics alignment settings, for each of the three tape speeds. The alignment-data memory is kept alive during power interruptions by an internal battery; tape counter and the 30-address program cue point data are not retained during power interruptions, however. This oversight would seem to be minor, as all sections of random-access memory could be easily serviced by the same stand-by power source.

The mechanics of the APR-5000 show vestiges of the MCI heritage. The MVC control is still a deck feature, although it mechanically resembles the Sony videodeck counterpart, relying on body capacitance sensing for activation in shuttle modes only. Head suspension is substantially identical to that of the JH-110 machines, with ISO metric hardware. The head assembly is rotated approximately 45 degrees counter-clockwise from the customary parallel-front position. Mercifully, the sheet of ½-inch aluminum that served as the deck chassis for MCI tape machines has been replaced with a true aluminum casting. Although still in use by at least one tape-machine manufacturer, the single-slab approach to deck design has been proven inadequate, and should have been universally abandoned by the industry years ago.

A spooling function is provided through use of the MVC control in shuttle mode. Winds are smooth and clean, although they do tend toward the uneven at the higher of the MVC speed range. Spooling does not engage the capstan.

Head-shield activation and tape-

Figure 5: Input monitor high-frequency group delay response. Curves are derived from the phase curves of Figure 4.

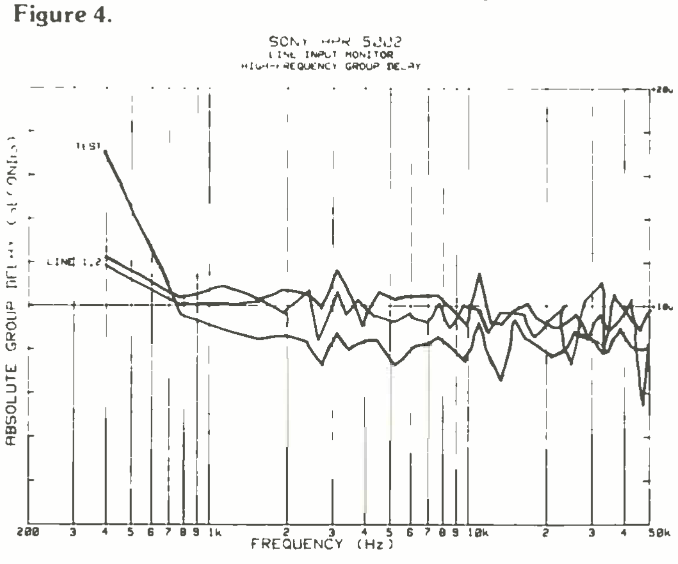
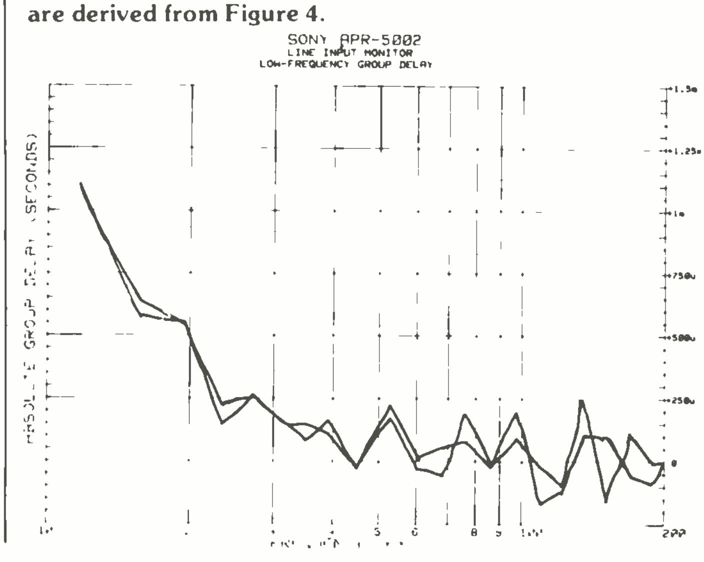
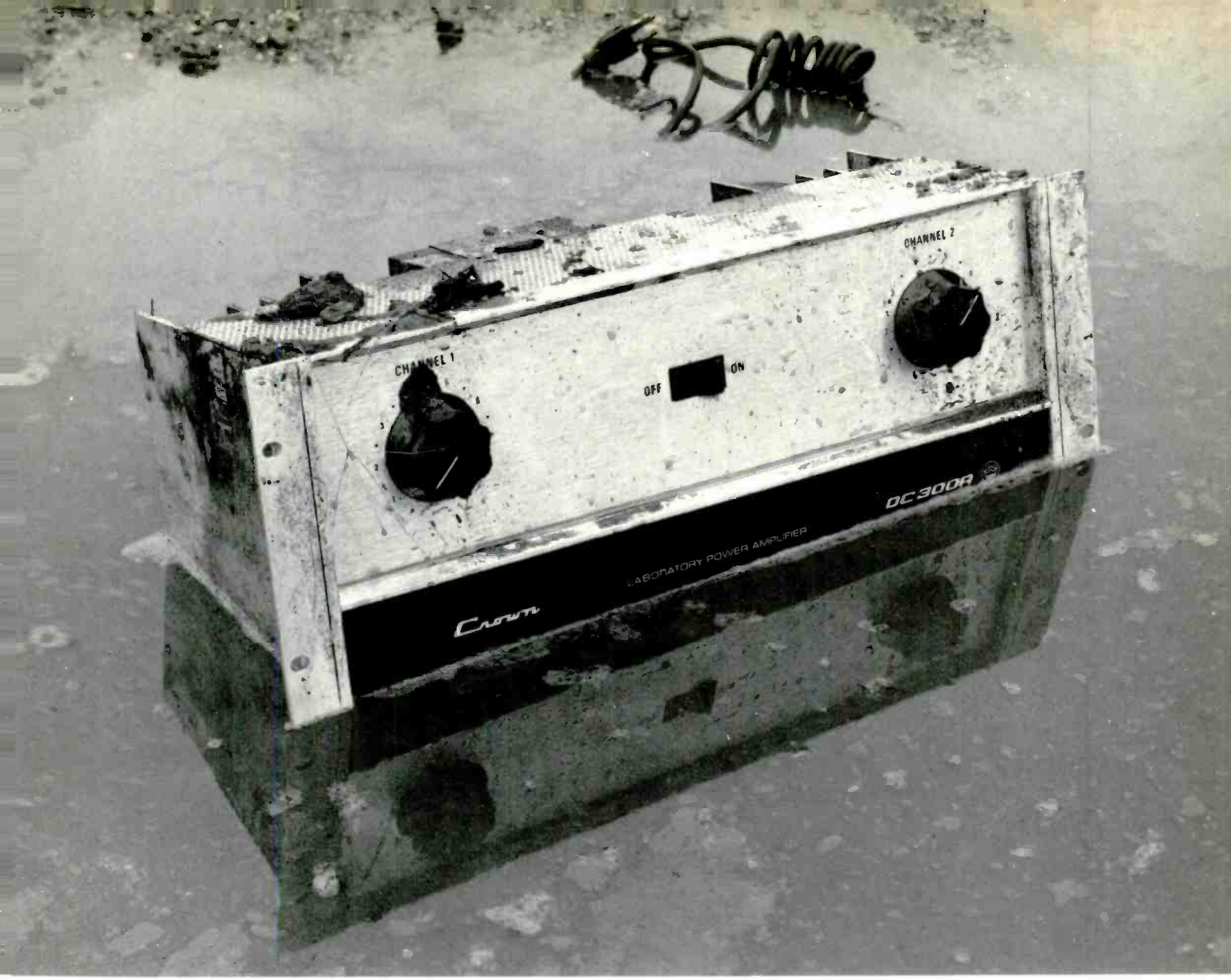


Figure 6: Input monitor low-frequency group delay response. Calibration Input/Output curve deleted. Curves are derived from Figure 4.





In the early evening of Sept. 17, 1973, Jay Barth was at the wheel of a 22 ft. utility truck that was loaded with sound equipment. Just south of Benton Harbor, MI an oncoming car crossed the center-line; fortunately Jay steered clear of the impending head-on collision. Unfortunately, a soft shoulder caused the truck to roll two and one half times. Exit several Crown DC-300A's through the metal roof of the truck's cargo area.

The airborne 300A's finally came to rest — scattered about in a muddy field, where they remained partially submerged for four and a half hours.

Jay miraculously escaped injury; the amplifiers apparently had not.

Unbelievably, after a short time under a blow-dryer all the amps worked perfectly and are still going strong.

The rest — and the truck, is history.

crown
1718 W. Mishawaka Road, Elkhart, IN 46517
(219) 294-5571

For additional information circle #94

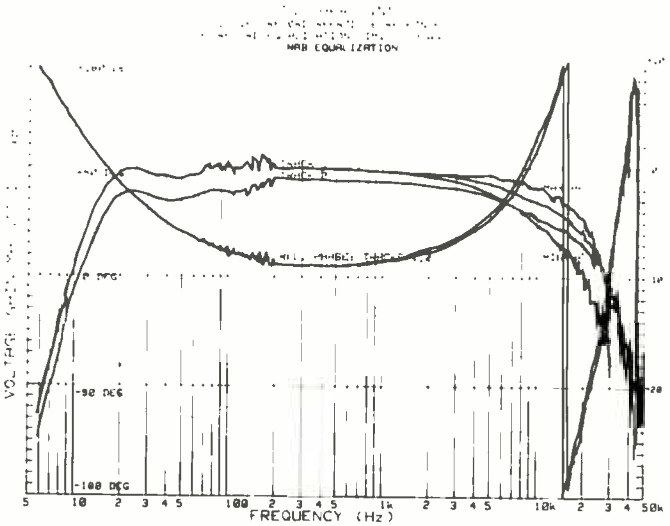


Figure 7: Record/reproduce transfer functions at 7.5 ips. Tracks #1 and #2 magnitudes are shown separated by 1 dB for clarity. The effects of maximum and minimum record HF boost are shown for track #1. Note that phase advances with frequency.

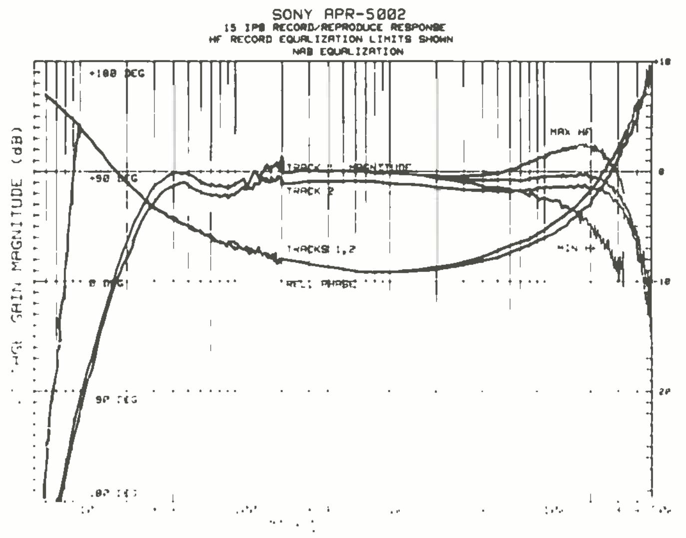


Figure 8: Record/reproduce transfer functions at 15 ips. The effects of maximum and minimum record HF boost are shown for track #1. Note that phase advances with frequency.

lifter functions can be defeated with toggling deck buttons. Changing the head-shield status does introduce a "thump" in the main audio channels, and should be avoided during play operation of the machine. Command of the machine can be transferred to an external controller and back by use of Network and Local command buttons. One should be able to safely presume that there is no interferance with Local manual operation of the deck from any external controller when the machine is set for Local control, and that the external controller is sheltered from deck status signals, which should be ignoring anyway when the APR-5000 is being manually controlled. Inasmuch as some machine controllers are not impervious to information that may be none of their business, this is a detail that can make a difference in convenience and efficiency of operations.

It is stated that local control of the deck is impossible in the Network mode. If this is absolutely true it could prove inconvenient, should a manual Stop command be necessary during external-control operations. Good sense dictates that the Stop command, from whatever source, should be able to override any other tape machine function, and that the user must never be denied his manual Stop option.

The RS-422 control port protocols and data timing schemes are not given in the manual. Attention is drawn to the possibility of the Sony Sync Master and BVE-5000 serving as machine controllers, but no promises are being made at this time; the relevant SMPTE documents governing the protocols are referenced, and that is all. The reluctance of most synchronizer and editing-system manufacturers to offer usable interfacing and control software for non-video devices is disheartening. It would be refreshing to encounter use of a standard data transfer and control system, such as the IEEE-488 standard, that would serve everyone's needs even though the overkill in using it would be extensive. It would seem that the tendency for equipment manufacturers to strive for new, unique and unfamiliar control interfaces, compatible only with a specific controller, is not in the long-term interest of anyone. I think it is time for the audio/video hardware industry to

Figure 9: Record/reproduce transfer functions at 30 ips. The effects of maximum and minimum record HF boost are shown for track #1. Note that phase advances with frequency.

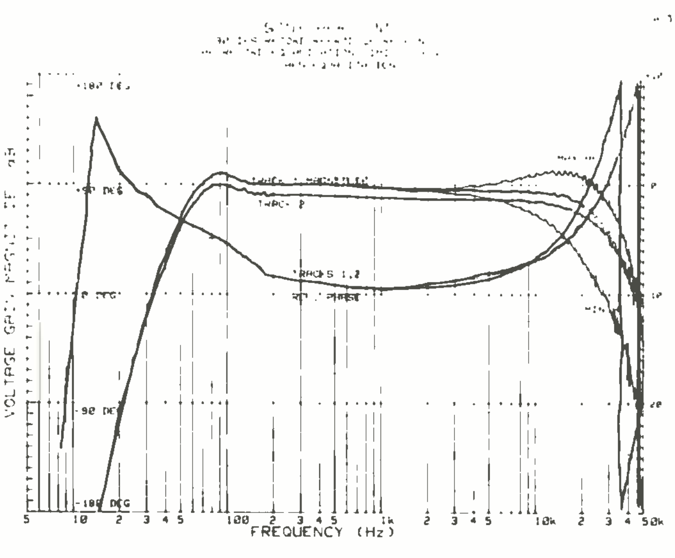
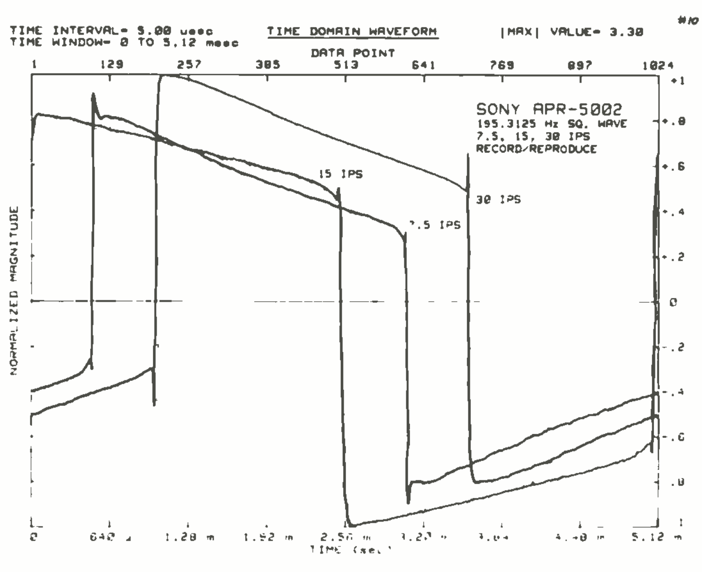


Figure 10: Time domain response at 7.5, 15, and 30 ips. The traces are displaced vertically and shifted in time for clarity.



stop re-inventing the wheel, and use technology that has served the rest of the electronics industry for over 10 years, and which has the capability of meeting its every possible need.

Spot erasure capability for either or both of the audio tracks, and perhaps for the timecode track, has been included as standard deck function. In this mode only the selected sections of the erase head are driven, so that manually-swept erasures can be accomplished without interference from any bias drive to the record head. If the timecode option does include this capability for the timecode channel, it should be used with great discretion since discontinuities in the timecode channel can render an otherwise perfect master unusable in synchronized applications.

The audio channels may be optionally muted during Lifter Defeat or other non-play operation by transmission of a FDR Start Enable command via the parallel control port or, presumably, the RS-422 port as well.

Triggering of external noise-reduction devices is possible by use of optically-coupled conductances between pins of a rear-panel connector provided for this purpose. These conductances are rated at 24 VDC in their "off" condition, and 100 mA in their "on" state.

Cue monitoring of the audio channels is provided through a small speaker located to the right of the VU-meter panel. Acoustic level should be adequate for most applications, and a ¼-inch stereo headphone jack is available.

Rear-panel BNC coaxial connectors provide access to the audio channels for a test generator and measuring instrument. Activation of the audio-channel alignment function connects the signal appearing at the Calibration Input BNC, so that it is added to the normal audio channel inputs to provide calibration drive to the record electronics. The output signal of the channel selected to be controlled by the computer is routed to the Calibration Output BNC for observation. I'm not entirely sure that this feature is of great value for a one-, two- or four-channel tape machine, although such a feature can be extremely helpful in multitrack checks and alignments. Phase (not polarity) relationships between channel outputs cannot be observed directly through the Calibration facilities, because only one output is observable at a time — although the vector sum of the two channel outputs should be available if "All" channel adjustment mode is selected while in Sync or Reproduce mode. Optimal azimuth alignment could then be determined by adjusting for a maximum magnitude indication at the Calibration Output port.

In the case of inclusion of a timecode option, it is not clear that a squared timecode signal would not be present also in the reproduce vector sum. Nor is it clear how the timecode bias, level, and equalization adjustments could be done without use of the extender card. Timecode and output will evidently be available as a retriggered, squared binary-level signal as seen at the timecode output connector.

Performance Assessment

The machine provided for testing featured transformerless audio signal paths. Audio-port inputs are of the single op-amp, differential amplifier-type, while the audio output ports are dual op-amp, quasi-balanced push-pull drive. Single-ended output connections may cause significant, non-destructive ground currents in the grounded audio lead, and may cause difficulties in some applications; caution is advised. Inclusion of transformers in the audio input and/or output ports is optional.

The main audio input ports look into a single op-amp differential input. Impedance versus frequency characteristics between audio connector pins 1 and 2, 1 and 3, and 2 and 3 are shown in Figure 1. The differential load measured 22.3 kohms at 200 Hz, greater than the 10 kohms claimed. The impedances between each side of the audio input pair and ground were

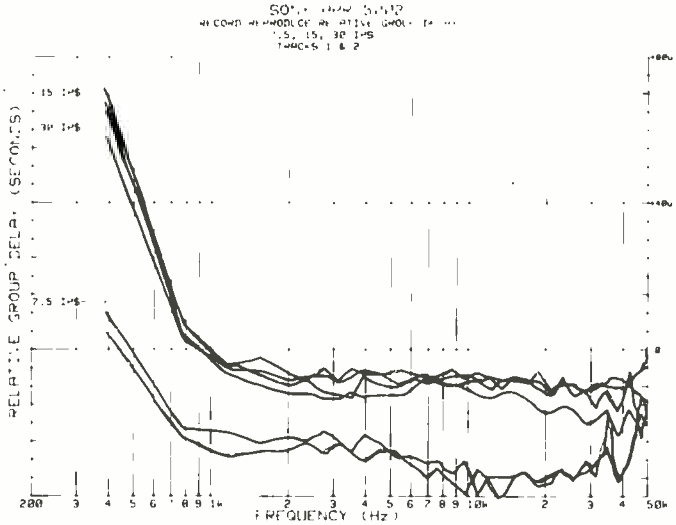


Figure 11: Record/reproduce relative group delay response. Group delays derived form Figures 7, 8, and 9 are shown overlaid.

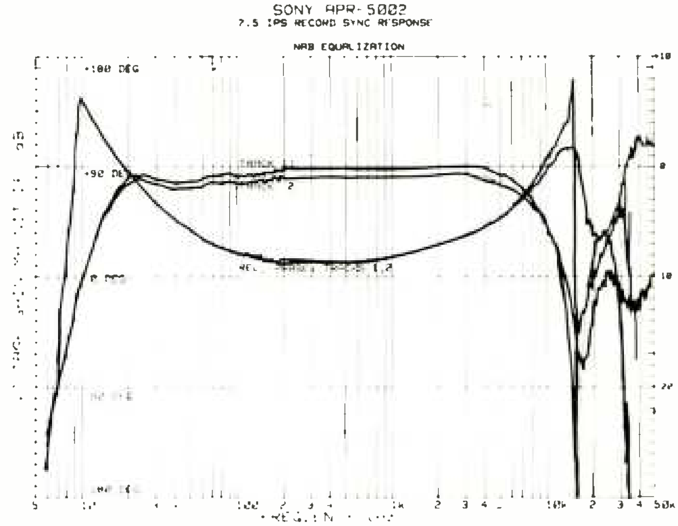


Figure 12: Record/sync transfer functions at 7.5 ips. Tracks #1 and #2 magnitudes are shown separated by 1 dB for clarity. The gap-loss null is readily observable in the magnitude null and phase reversal circa 15 kHz. Note that phase advances with frequency.

measured at 12.5 and 9.17 kohms, an imbalance of about 4:3. Inclusion of R-C lowpass filtering and cable capacitance is likely responsible for the drop in impedance magnitude at frequencies above about 2 kHz. Normal mode impedance magnitude declines to about 10 kohms at about 60 kHz, while the imbalance side impedances bottom out at about 3.8 kohms at the same frequency.

Input port common-mode performance is shown in the photograph of Figure 2 as observed at the respective main audio channel output ports. The top trace is the normal mode response at +4 dBv, channels 1 and 2 overlaid. The center trace is channel 1 common-mode response, and the bottom trace is the channel 2 common-mode response. Channel 1 and 2 CMRR is seen to be about 45 and 62 dB, respectively. Input port interfacing should present no problems.

Figure 3 shows the output-port impedance magnitudes for channel 1 output typical. Normal mode source impedance measured 128 ohms at 200 Hz, remaining constant to about 90 kHz. Line imbalance is about 3.62/3.23 or 9:8 for a perfectly floating load. I would like to see an output balance ratio closer to 1:1 for balanced outputs, although this condition seems to serve well enough. In situations where an active-balanced output is driving an imbalanced differential input (the most common condition), the resulting degree of line balance is determined by the parallel combination of side impedances of each port from signal pair to ground. If the output port has a fairly low impedance magnitude of side impedance, and is closely balanced, the condition of the entire circuit will tend to be in balance, even though the differential input load is imbalanced when observed by itself. This can be a significant matter in the case of high-intensity RF and magnetic fields where transmission line imbalance is most apparent. A common-mode rejection ratio of about 18 dB could be expected for a line imbalance of this magnitude.

Input-monitor transfer functions for the APR-5002, surprisingly, were found to be inverting for the normal audio channels. The Calibration Input proved to be the exception, having an erect polarity (not phase) response as observed at the Calibration Output port and at the main audio channel output connectors, pin 3 taken as "HI." Audio-connector pin 3 was taken as the "HI" signal terminal for

SUMMARY OF SONY APR-5002 SPECIFICATIONS

Specification	Quoted	Observed
Power Requirements:	102/120/138/ 204/240/276 VAC; 48 to 60 Hz.	Yes
Power Consumption:	300W max.	Not checked.
Fuse Rating:	5A (102V), 4A (120V) 2A (240V); normal or fast blow.	Yes
Reel Size:	3 to 12.5 inches; NAB or EIA hubs; DIN hubs optional.	Yes
Tape Format:	0.25-inch, two-track NAB track-standard; 0.25 inch, two-track DIN track standard; 0.25 inch, 3-track center-track timecode. (0.5 inch, two-track timecode feature optional.)	As tested.
Tape Speed:	Standard (high-speed) 7.5, 15 and 30 ips; optional (low speed) 3.75, 7.5 and 15 ips; ±50% variable.	As tested.
Speed Stability:	Better than 0.02%.	±0.017% head to tail; 2,490 feet.
Start-Up Time	30 ips 900 mS; 15 ips 500 mS; 7.5 ips: 500 mS.	Not checked.
Fast-Wind Time:	2,400 feet 110 sec; 4,800 feet 170 sec;	2,490 feet 114 sec; Not checked.
Spool-Wind Time:	2,400 feet 370 sec;	2,490 feet 387 sec.
MVC Velocity:	0 to 1.9 meters per sec.	Not checked.
Wow and Flutter:	30 ips less than 0.025% 15 ips less than 0.035% 7.5 ips less than 0.055% 3.75 ips less than 0.10% (±peak, DIN-weighted).	30 ips: ±0.017% 15 ips: ±0.027% 7.5 ips: ±0.037% 3.75 ips: Not Tested W+F, peak, DIN-weighted.
Frequency Response: *Record/Play*	**30 ips:** 40 Hz to 28 kHz; +0.75/-2 dB.	**30 ips:** 53 Hz to 20.9 kHz; +0.8/-2 dB.
	15 ips: 30 Hz to 24 kHz; +0.75/-2 dB.	**15 ips:** 29 Hz to 30.0 kHz; +0.1/-2 dB.
	7.5 ips: 30 Hz to 20 kHz; +0.75/-1.5 dB.	**7.5 ips:** 14 Hz to 5.27 kHz; +1.3/-1.5 dB.
Frequency Response: *Record/Sync*	**30 ips:** 50 Hz to 16 kHz; +0.75/-2 dB.	**30 ips:** 42 Hz to 25.6 kHz; +1.4/-2 dB.
	15 ips: 30 Hz to 10 kHz; +0.75/-2 dB.	15 ips: 27 Hz to 21.3 kHz; +0/-2 dB.
	7.5 ips: 30 Hz to 4 kHz; +0.75/-2 dB.	**7.5 ips:** 15 Hz to 6.8 kHz; +0/-1.5 dB.
Recorded (S+N)/N Ratio: Record/ Reproduce	30 ips 66 dB. 15 ips 64 dB. 7.5 ips 63 dB. 20 Hz to 20 kHz UNWTD; Ref: 510 nW/m.	30 ips 66.5 dB. 15 ips 62 dB. 7.5 ips 57.5 dB.
Recorded Distortion: (1 kHz; 510 nW/m) *3rd Harmonic*	30 ips less than 0.35%; 15 ips less than 0.52%; 7.5 ips less than 1.6%.	30 ips 0.16% 15 ips 0.50% 7.5 ips 1.8%.
2nd Harmonic	30 ips less than 0.10%; 15 ips less than 0.10%; 7.5 ips less than 0.10%.	30 ips: 15 ips: 7.5 ips: Unreadable at 30 Hz; analyzer resolution.
3% 3rd Harmonic Fluxivity Level:	30 ips: 1,040 nW/m. 15 ips: 1,020 nW/m; 7.5 ips: 1,000 nW/m.	Not measured.
Bias Frequency:	400 kHz.	Not measured.
Erase Frequency:	100 kHz.	Not measured.
Depth of Erasure:	Greater than 76 dB below 1 kHz at 250 nW/m; all speeds and track formats.	Erasure below 1 kHz at 520 nW/m: 30 ips 78/82 dB; Tk 1/2. 15 ips 80/84 dB; Tk 1/2. 7.5 ips 83/83 dB; Tk 1/2.
	Audio Amplifier Electronics:	
Input Impedance	10 kohms, balanced.	22.3 kohms, imbalanced differential input.
Output Impedance	120 ohms, balanced.	128 ohms, imbalanced active differential.
Maximum Audio Output level	Not quoted.	+22.4 dBv (600 ohms); IHF load.
Weight *Table Top*	91 pounds; 46.26 kg.	Not measured.
Stand type	110 pounds; 49.89 kg.	Not measured.
	ADDITIONAL OBSERVED:	
Polarity Response:		Audio port pin #3 taken as "HI":
Record/play		Negative
Record/Sync		Negative
Line In/Out		Inverting
Generator In/Meter Out		Non-inverting
Generator In/Line Out		Inverting
Line In/Meter Out		Non-inverting
Repro/Meter Out		Inverting
Sync/Meter Out		Inverting
Hardware:		ISO Metric: Hex socket-head; Totsu head ("slot 'n' dot")
Testing tape stock:		3M 226-½
Flux-Frequency References:		MRL 21L221, 21J205, 21T204.

Suggested End-User Price: APR-5001 (Mono) $6,800; 5002 (DIN/NAB stereo) $7,500; 5003 (DIN stereo plus IEC timecode) $9,000; SU-14 stand $500.

Manufacturer: Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. (201) 930-6137.

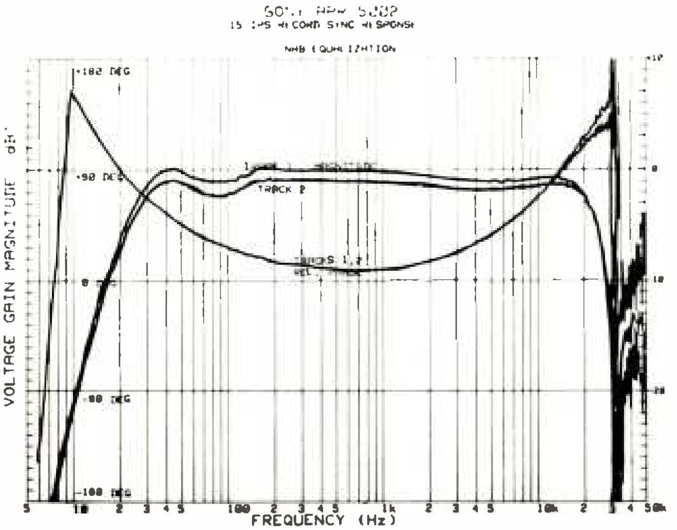


Figure 13: Record/sync transfer functions at 15 ips. Tracks #1 and #2 magnitudes are shown separated by 1 dB for clarity. The gap loss null is readily observable in the magnitude null and phase reversal circa 33 kHz. Note that phase advances with frequency.

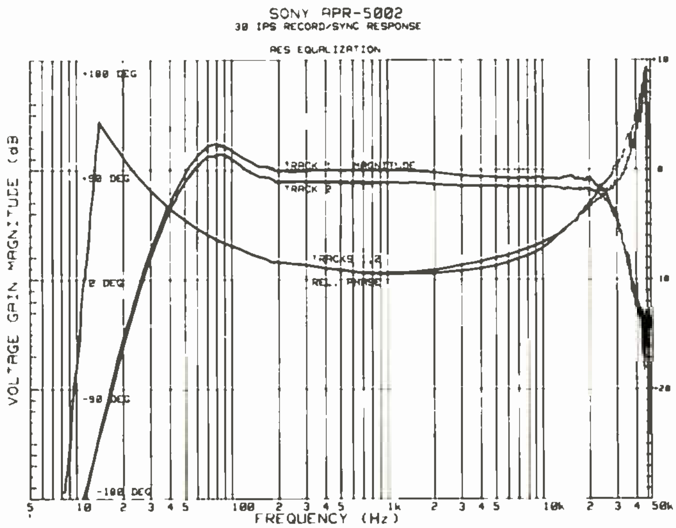


Figure 14: Record/sync transfer functions at 30 ips. Tracks #1 and #2 magnitudes are shown separated by 1 dB for clarity. Note that phase advances with frequency.

all measurements given here, which conforms with the conventions followed by both Sony and MCI in the past.

The input-monitor transfer functions are shown in Figure 4. Phase (not polarity) curves for main audio channels 1 and 2 are displaced -180 degrees so that they will tend to rotate about the central abscissa of the graph. The line-input magnitude functions are remarkably flat at low frequencies, showing only about 10 degrees of lead at 6 Hz. By contrast, the Calibration Input/Output response is down about 3.5 dB at 6 Hz with a phase lead of around 53 degrees, and down about 0.5 dB at 20 Hz. It should be possible to improve this performance to match the main audio channel response. At the high-end of the band, the main audio channel 1 magnitude tends to roll off a bit faster than the channel 2 or Calibration Input/Output does. All three channels were found to exhibit very linear lagging phase response with increasing frequency.

Group-delay response for the APR-5002 main audio and Calibration Input/Output channels is shown in Figures 5 and 6. Figure 5 shows that, in all cases, the curves flatten out and are approximately horizontal to 50 kHz, indicating equal propagation delay at all frequencies in that range. The notable exception is the delay curve labeled "Test" which rises steeply below frequencies below about 800 Hz — is a consequence of the low frequency magnitude rolloff in the Calibration channel noted above. Low-frequency group delay is shown in Figure 6 with the Calibration delay curve deleted, as it is off-scale through most of this region. The main audio channels' group delay becomes fairly flat above about 60 Hz, and oscillates about the zero-second abscissa above that frequency — a remarkably good performance for any AC-coupled audio signal device.

The APR-5002 attends to its prime task of recording and reproduction very well indeed, as witnessed by the curves of Figures 7, 8 and 9. The low-frequency contour irregularities are restricted to a band +1.5 -1 dB about the magnitude reference. Record/reproduce magnitude response was adjusted to be flat at 1 and 10 kHz against the calibrated reproducer using the CW sinewave signal. Flatness was observed to be within -1 dB of that reference line between the two frequencies. The data shown is the result of a deconvoluted Fourier transform of a 195.3125-Hz squarewave, and therefore represents a more dense signal spectrum than would a sinewave sweep. The magnitude response limits given in the Summary of observed performance were determined from these curves.

A seasoned reader of these equipment assessments will note the unusual signal phase-function behavior shown in all of these record/reproduce graphs. In this case, the signal phase *leads* rather than lags with increasing frequency — as is more often encountered for electronic signal paths in general — due to the inclusion of a variable, single-pole allpass filter network in the record electronics signal path. The frequency of the allpass pole is adjusted by the computer to track the phase response of the record pre-emphasis filter. This can be done with reliability, because there is no phase contribution to the analog record/reproduce processes, leaving only the electrical responses of the various electronic stages involved to be contended with. The reproducer phase response is largely dictated by the time constants of the reproducing characteristic to be used, leaving only the record phase response as the major variable in the system.

Time-domain plots of track #1's 195.3125-Hz squarewave response are shown in Figure 10. Each of the displaced squarewave traces shows a pre-transition spike, which is fairly small at the 7.5 ips speed, growing somewhat at 15 ips and becoming quite large at 30 ips. The 7.5 ips waveform shows a post-transition spike, implying that as great as the induced corrective phase lead may be at that speed, it is still not adequate to minimize the overshoot due to the phase rotation caused by the radical HF boost required to overcome the tape-wavelength response for flat magnitude response at that speed. At 15 ips, the phase correction would appear to be about the correct value for minimal overshoot, while the phase correction for the 30 ips case appears to be more than required. The absence of any ringing in the region of the squarewave transitions of Figure 10 indicates very accurate reproducer head termination.

The APR-5002 record/reproduce, relative group-delay curves for tracks #1 and #2 are shown in Figure 11. The conclusions drawn from examination of the character of the time-domain traces of Figure 10 are confirmed for the cases of the 7.5 and 15 ips speeds. The slight drop in the 7.5 ips delay in the region 6 to 30 kHz does look like the delay compensation could profitably be increased in this region. The 30 ips curve is close to optimal, with a slight droop above about 10 kHz. The choice of delay compensation as shown here is very close to optimal, and likely could not be improved very much to yield truly flat group-delay response.

The flux-loop sweeps of Figure 15 show the linearity of the pre-equalized reproduce head response versus frequency, while Figure 16 shows the same kind of data for the sync head. The lack of any deviation from linearity about the peak due to undamped head resonance is clearly indicated, showing careful attention to critical termination of the respective track cores. The two channels are very closely matched in all cases.

The flatness of the APR-5002 metering is quite good; Table 1 gives the observed meter response for a constant main audio input level. Examination of the magnitude curve for track #1 given in Figure 4 shows that the meter follows the channel magnitude response closely over the range measured.

Analysis Summary

The APR-5002 is easily the best analog tape machine to emerge from any manufacturer over the past several years. The quality of performance exhibited in these tests indicate that there may be hope for high quality audio recording in the absence of any more Ampex ATR-100s, the benchmark against which I have tended to judge all other analog machines. The wow and flutter performance is very good, although there is a tendency for the flutter readings to rise above their lowest values at the extreme head and tail of the reel.

The service manual is very complete, and offers step-by-step procedures for most significant repair and maintenance actions. The schematics and parts lists show the same completeness that Sony has given to its video equipment. Once one is introduced to the documentation philosophy and the nomenclature, it is fairly easy to find the relevant section of the manual that applies to a given situation.

The Totsu (slot 'n' dot) hardware may present problems in tooling for the technician accustomed to dealing with slot, hex-socket, Phillips and Posidrive hardware. Yielding to the temptation to make do with careful application of a common slot-blade screwdriver rather than the proper Totsu tools will result in unnecessary mayhem. The investment in proper quality tooling is always a rewarding, if intangible, one.

The machine is small, light and easily moved. All major electronic assemblies are accessible from the front, or by removal of the rear panels. The audio mother board is more than just a carrier of interconnects and board receptacles; it contains a couple of data demultiplexers, LED drivers and the cue-speaker drive amplifiers. The placement of significant active components on relatively inaccessible mother boards is, I think, a poor choice of configuration from a tech support viewpoint. The obvious difficulty in observing a circuit contained on a mother board assembly can be frustratingly since installation of the daughter boards is often required to

TABLE 1: VU METER INDICATION VERSUS FREQUENCY. (Constant line input.)

Frequency	Meter Indication
5	0.0 (dB)
10	0.0
20	0.0
50	0.0
100	0.0
250	0.0
500	0.0
1 k	0.0
2 k	0 0
4 k	0.0
6 k	0 0
8 k	0.0
10 k	-0.05
12.5 k	-0.15
16 k	-0.25
20 k	-0.35
25 k	-0.60
30 k	-0.80
40 k	-1.65

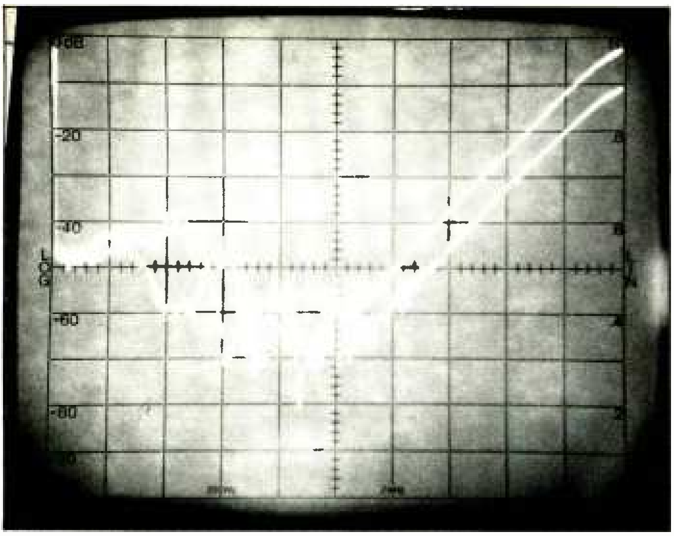


Figure 15: Reproduce head flux-loop response. The upper trace is track #1 response, track #2 is the lower trace. Scale factors are 10 dB/division, vertical, and log frequency, horizontal, as graduated.

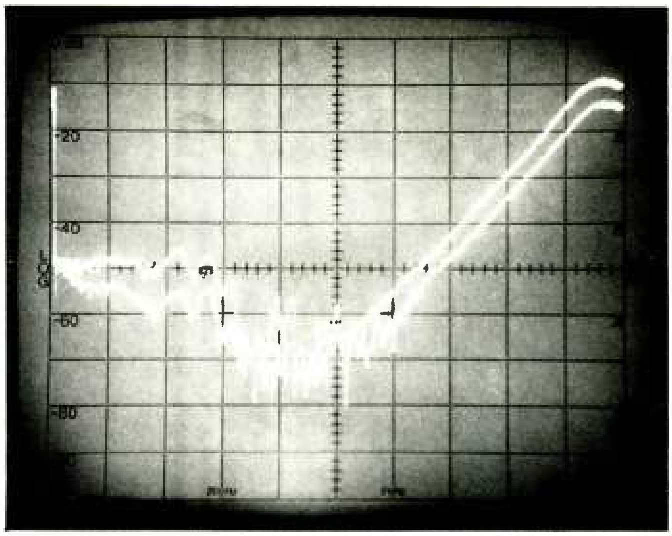


Figure 16: Sync head flux-loop response. The upper trace is track #1 response, track #2 is the lower trace. Scale factors are 10 dB/division, vertical, and log frequency, horizontal, as graduated.

place the system in a configuration permitting intelligent fault isolation and observation of dynamic signals.

The mechanical design is rugged, and the many exploded assembly drawings clear and uncluttered. Printed-circuit assemblies show the Sony influence, and bear white silk-screened component designations over a solder-resist coating. (There are a couple of "Oh yeahs" evident as components tacked onto foil runs in places they were obviously not originally intended — these problems will undoubtedly be corrected as serial numbers advance.)

The polarity-inversion problem is just one of those things that sometimes slip through the engineering design review process and, though mundane, needs to be corrected immediately. The most advantageous solution to this problem would seem to be reversal of the audio pairs at the main audio-input connectors, which would bring all signal ports into polarity agreement and result in erect recording and reproduction in both sync and play modes.

The matter of the inverse hexadecimal parameter indications is more a matter of preference on my part than a deficiency. The hex values are unique to any parameter magnitude, and serve their intended purpose.

The phase (not polarity) correction network seems to be designed very close to optimum for the speeds and equalizations observed. The ability of the microprocessor to track the record equalization delay as closely as it does is a compliment to its designers. Removing responsibility for proper delay compensation from the back of the maintenance technician is a definite technical step forward.

In closing, I should think that a deck having 14-inch reel capacity and four-track channel capacity would be a worthy entry for the pro audio/video market. Four-track is still widely used, since the central timecode track convention is not yet widespread, and new equipment capable of format compatibility with existing program libraries is still needed. ■■■

MANUFACTURER'S REPLY

Hiro Konno and Takeshi Yazawa, of the Product Management department, Professional Audio Division, Sony Corporation of America, reply as follows:

We appreciate the author's thorough evaluation and his report on our new product, the APR-5002. We have found it to be quite detailed, and our consensus is that the author did an excellent job.

However, there are some discrepancies in his findings, some misinterpretations of design goals, and general differences of opinion. We gratefully acknowledge the author's pointing out the polarity reversal in the audio path in production units. This problem has since been resolved by Engineering.

The areas requiring clarification are as follows:

1. **Published versus Measured Specifications Discrepancies:**

In this particular area, it appears as though the author's measured specifications were based on criteria other than those used by Sony engineering. Therefore his data is slightly different. Some of our specifications have recently been ammended (S/N ratio, 30 ips 65 dB; 15 ips 62 dB; 7.5 ips 62 dB), actually eliminating some discrepancies.

We do find a large variance, however, in his measured 7.5 ips frequency response and S/N and our factory measurement. We are not in a position to comment further without consulting the author on his test procedure and problems encountered. [We understand that Peter Butt used a 195-Hz squarewave excitation signal to provide frequency response measurements, which are available by examining the time-domain traces thus produced — *Editor.*]

2. **Timecode Processing:**

As this evaluation was intended to be on an APR-5002, we chose to provide no description of our timecode processing methods, or any actual operational specifications. The APR-5003 is uniquely different from the APR-5002, and trying to lump them together in an evaluation based on the APR-5002 alone would not be appropriate in our opinion. We are very pleased to supply an APR-5003 at a later date for a complete evaluation on its own.

3. **Memory Mapping and Data Back-Up:**

The APR-5000 Series retains all Audio Alignment, Search Cue points and Timecode status information in a battery backed-up memory system. Each specific type of data is mapped into its own unique locations, and is only written over by an operator performing a detailed alignment procedure. This data back-up also allows for each of the 12 possible head assemblies to have its own data for call-up at any time in the future. This data is also supported in the case of the power supply being reviewed for service. We do not understand why the author may have had a problem in this area.

4. **Serial Control Protocols:**

This is an area of conjecture by almost everybody today. We have chosen to support two protocols in the APR-5000 Series, both of which can only be practically realized in the

APR-5003, using a Time Referencing System (i.e. Timecode SMPTE/EBU). They are the Sync Master specific, RS-422A and specific sub-set of the Sony BV-type, emulating a BVU-800 for Transport Motion Control Commands. We refer to the work of SMPTE/EBU and the published documents to date in the manual. This is meant to give the readers some insight as to the work going on for the future implementation. In respect to protocol listings, Sony has a policy of supplying them on a controlled basis to qualified users, but not publishing them outright.

5. **Line Input Differential Coupling Imbalance:**
We do not agree with the author in his evaluation of this section at all. We use a hybrid Diff Amp for this section and, upon examination of the actual circuit, one can see that his results are not possible. We feel that he may have incorrectly connected his test equipment to the Input Amp and unbalanced it improperly, resulting in the results mentioned.

6. **Author's Comments on Signal Processing Performance:**
We feel that these are based upon personal opinion and, although valid on a strictly theoretical basis, are not achievable with the technology that is available today.

As always, we will continue to endeavor in achieving absolutely "perfect" audio performance, as technology allows.

7. **Audio Mother Board Servicing:**
The author comments on the lack of serviceability of this particular PCB. In actuality, this is *not* the Audio Mother Board, but the Audio Control Mother Board, located in the Meter housing. This PCB is easily accessed and serviced by anyone. In addition, the real Audio Mother Board is a passive backplane, which can be removed and repaired with the same relative ease.

8. **Polarity Reversal:**
The description of the reversal in polarity of the Audio Signal Path is correct and incorrect. He assumes that we use pin #3 as "HI," and proceeds to describe the reversal in terms of input coupling problem. In actuality, we conform to IEC Standard #286, which calls for pin #2 to be the "HI" in XLR connectors. The conclusion that we came to is that the coupling problem is actually in the output path (where it was later actually found and corrected).

As time does not permit consultation with the author prior to publication, we are confident that *R-e/p* will afford us the opportunity to reply in greater depth following our dialog.

□□□

New Products

MODEL 246 PORTASTUDIO COMBINED MIXER/CASSETTE RECORDER FROM TASCAM

A portable, completely self-contained, four-track production system the 246 Portastudio is said to successfully meet the recording needs of artists involved in music, electronic music, multi-image, and non-sync video. As a complete system, the 246 combines a full function 6×4×2 mixer, and a two-speed, four-track recorder with dbx noise reduction.

In addition to portability, the 246's all-in-one packaging provides automatic switching and single-step switching, virtually eliminating time consuming and confusing patching necessary on many multitrack recorders and mixers. Automatic monitor switching during punch-in and insert recording, one-button meter and 'phones switching, and tape-signal routing for remix, also are featured.

Two-speed operation and a dbx defeat switch allow tapes made on other Tascam multitrack cassette machines to be played on the 246, and vice-versa. Memory transport functions and a Zero-Return allow sections of the tape to be repeated automatically for rehearsing punch-ins, overdubs or remixes. The Portastudio may also be controlled by the RC-71 Remote Transport Controller or the RC-30P controlled by the RC-71 Remote Transport Controller or the RC-30P Remote Punch-in Footswitch.

Any of the six mixer channels may be recorded by any or all of the four tracks of the recorder at any time, via a combination of bus. Assign and Record Function switches. A stereo effects system and stereo monitor mixer system are also provided. In addition to the four meters that are automatically switched to display the signal level in the four busses or four recorder tracks, the 246 also has a pair of meters that can be switched to read the stereo bus, monitor outputs, or the effects outputs.

The Model 246 Portastudio has a suggested retail price of $1,300.

TASCAM

For additional information circle #100

AURATONE MODEL 5MC RACK MOUNTABLE THREE-CHANNEL MONITORS

Designed specifically as an ultra-compact, three-channel monitor for the video/broadcast industry, the Model 5MC is equivalent to three, side-by-side 5C Super-Sound-Cubes contained in a single enclosure measuring only $5\frac{1}{2}$ by $16\frac{1}{2}$ by $8\frac{1}{2}$ inches. The unit will provide separate audio feeds for functions such as cue, program, emergency channel, talkback, news, sports, etc., and can be used for A-B comparisons of stereo-mono quality for both broadcast program material, and in recording production. In this latter application, the two outside channels are used for stereo, with the center for mono.

Quoted frequency response for the full-range close-field monitors is ±3.5 dB on-axis, (anechoic from 150 Hz to 12.5 kHz). Each five-inch round driver has a specially shielded magnet structure for preventing distortion of the TV image on adjacent CRTs due to flux leakage. Impedance is 8 ohms, and power handling 30 watts per channel.

Optional metal rack ears simplify mounting in standard 19-inch relay racks. Protective plastic bumpers are provided for use on console meter bridges or desk tops and nylon hangers are included for wall mounting.

Suggested pro-net price for the Model 5MC Multi-Channel Monitor is $150; an optional 19-inch rack-mounting kit is $10.

AURATONE CORPORATION

For additional information circle #101

GARFIELD ELECTRONICS MASTER BEAT FOR TIMECODE/MIDI SYNCHRONIZATION

The new unit provides sync generation in beats per minute, 24, 25, or 30 fps film/video calibrated tempos from all SMPTE/EBU formats: 24- and 25-frame, 30 DF and NDF, and simultaneous production of 16 musical-instrument sync formats. In addition, a "Doctor Click" facility enables sync to click tracks, live tracks, MIDI, and all tape sync codes.

A high-resolution, programmable "timing map" allows any beat-interval sequence to be stored, edited, and offset under SMPTE control, for adjustment of rhythmic "feel." Also available are six multi-programmable SMPTE controlled-event gates, each with five-volt and contact-closure outputs for synchronized sound effects and control triggering.

Other production features include RS-232 serial interface; code regenerate and conversion; jam sync; SMPTE from Neopilot; remote control-inputs; and non-volatile memory.

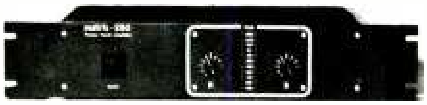
Recommended list price of the Master Beat is $1,995.

GARFIELD ELECTRONICS

For additional information circle #102

HOQUE ELECTRONICS UNVEILS MATRIX M-250 POWER AMP

The Matrix M-250 stereo power amplifier will provide 100 watts rms into 8 ohms and 180 Watts rms into 4 ohms, with less than 0.05% distortion, and features extensive amplifier and load protection plus has precision peak-power LED meters with clip indicators.



A high-speed, non-switching design is said to provide an accurate wide-band response of 5 Hz to 100 kHz, with less than 120 dB signal-to-noise ratio.

Two toroidally-wound transformers permit the amplifier to be housed in a 3½-inch rack mounting steel cabinet, and be powered by 115 or 230 line voltage.

Suggested pro-user price for the Matrix M-250 amp is $458.

HOQUE ELECTRONICS

For additional information circle #103

SHURE ANNOUNCES TWO NEW BOUNDARY-EFFECT CONDENSER MICROPHONES

The two new additions to the company's line of boundary-effect (surface) microphones are the Model 819 Unidirectional and the Model 809 Omnidirectional condenser models. Both mikes, according to Sandy Schroeder, Shure's marketing manager professional entertainment and general audio products, "These two new microphones will offer sound contractors and music dealers improved performance, durability, and flexibility in those applications where a boundary-effect microphone is required, but where budget is limited.

Both microphones are priced below that of the professional Shure SM90/SM91 models and should prove to be very attractive to the general sound and semi-professional markets, because of their superior reproduction quality and long-term reliability."

Both the 819 Unidirectional and the 809 Omnidirectional Condenser Surface Microphones have a user net price of $200.

SHURE BROTHERS, INC.

For additional information circle #104

New Products

SCIENTIFIC DESIGN ANNOUNCES NEW COMPUTER-AIDED SPEAKER DESIGN SOFTWARE

The Computer-Aided Speaker Design, for use on an Apple II computer, was developed for contractors, sound companies, consultants and manufacturers. This package allows complete modelling of loudspeaker systems, and accurately predicts system response before any prototypes are built.

Version 2.0 now comes as a two-disk set. Besides the main program disk, a file disk allows data storage on up to 800 speaker drivers, and is shipped loaded with data on over 120 drivers from various manufacturers. The program calculates and graphically plots the response and displacement limited functions for both sealed and vented systems. The inclusion of an electronic filter can also be incorporated in the design chain. The user has full control over graph scales and resolution.

A complete crossover design program allows 6, 12, 18 and 24 dB per octave crossovers to be designed complete with impedance correction circuits. Nine utility programs calculate various often-used functions, including reference efficiency from Thiele/Small parameters, vent tunings, passive radiator mass, etc.

Computer requirements are an Apple II computer with 64K memory; Applesoft ROM and two disk drives. The Grappler parallel printer interface is supported for graphic screen print-outs. An IBM version is presently in development.

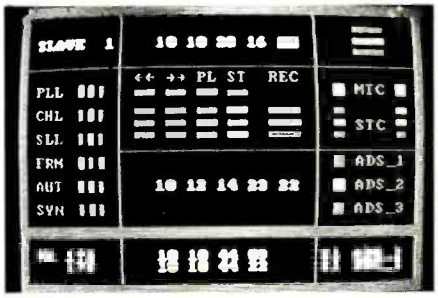
Computer-Aided Speaker Design has a suggested retail price of $99.95.

SCIENTIFIC DESIGN SOFTWARE

For additional information circle #109

SOUNDMASTER AUDIO EDITING SYSTEM FROM AMTEL SYSTEMS

SoundMaster is a SMPTE timecode-based, computer-assisted editing control system, designed for audio-for-video and audio-for-film post production, and is described as offering the speed, ease of operation and flexibility previously available only in videotape editing systems.



The system can simultaneously control up to four tape machines (typically three audio and one video transport), using tape synchronizers as intelligent interfaces to the transports. An IBM microcomputer acts as a host for SoundMaster's custom hardware and software.

The system can store and execute over 2,500 editing events. Three different video display screens are generated that furnish the user with: real-time transport status display, an edit decision list with list management functions; and a system parameter set-up screen. [See a feature article in the June issue of *R-e/p* for a full description of operational features in-use at The Master's Workshop —*Editor*.]

List price of the Sound Master system is $9,995, which includes the IBM computer. Users who already own the necessary computer hardware may purchase the system for $8,195.

AMTEL SYSTEMS, INC.

For additional information circle #110

E-MU SYSTEMS ANNOUNCE EMULATOR SP-12 SAMPLING PERCUSSION SYSTEM

The new unit uses 12-bit data sampling, and comes pre-programmed with a full complement of acoustic and electronic drum and percussion sounds. Additional sounds can be loaded from cassette, an optional disk drive, or custom sounds recorded with the SP-12's built-in user sampling facility. Because the SP-12 samples into battery-backed memory, sounds are always available instantly the SP-12 is turned on.

Tuning, decay, mix level and tempo (including gradual accelerados and de-accelerados) are totally programmable, while touch-sensitive play buttons provide control over dynamics and expression. And with the SP-12's multiparameter modes, any sound can be assigned to all eight play buttons

For additional information circle #108

simultaneously with a different tuning or level setting for each button.

Flexible repeat and subsong functions greatly simplify the creation of complex sound structures. A step programming mode lets the user create intricate rhythm patterns, as well as examine and edit patterns originally programmed in real time.

For use in professional film, video, and recording applications the SP-12 includes a built-in SMPTE code reader/generator, as well as a standard 24 pulse-per-quarter clock. A full MIDI implementation allows control from MIDI keyboards, sequencers, and computers.

Suggested list price for the Emulator SP-12 is $2,745.

E-MU SYSTEMS, INC.

For additional information circle #111

SENNHEISER ELECTRONIC INTRODUCES EM 2003-9 NARROWBAND VHF RECEIVER

Utilizing true diversity circuitry, the new receiver is said to offer consistent, noise-free performance. Available for use with carrier frequencies ranging from 150 to 216 MHz, the EM 2003-9 offers uninterrupted, broadcast-quality audio, and employs the same front-end helical filters as the company's larger multi-channel diversity receivers, which result in extremely high selectivity of RF, and the elimination of intermodulation from other stray carriers.

The unit may be rack-mounted or stacked, and has an integral antenna splitter for efficient use of antennas and installation simplicity. LED instrumentation displays both audio modulation and RF strength, as well as the selection of the strongest RF signal at the two antennas. Adjustable squelch guarantees positive muting of unwanted signal. Two discrete outputs — balanced and unbalanced — are available, each with a separate level control for applications requiring both monitor mix and live mix without the use of line splitters.

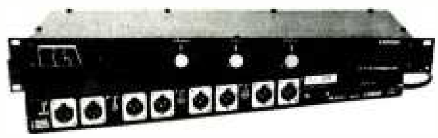
SENNHEISER ELECTRONIC CORPORATION

For additional information circle #112

New Products

RLA INTERNATIONAL INTRODUCES X3000A ELECTRONIC CROSSOVER

The two-channel, three-way unit provides a variety of outputs: a sub-bass output with a bandwidth of 30 to 100 Hz; a full output with a bandwidth of 30 Hz to 65 kHz; and a tweeter output with a bandwidth of 7 to 65 kHz. The sub-bass and tweeter outputs have front-panel level controls allowing 16 dB of boost or cut, while the full output has a level control allowing 6 dB of boost or 40 dB of cut. The input signal is thus divided into a buffered and subsonic-filtered, full-spectrum output that is available to feed the main system crossover, as well as two special purpose low- and high-end drive signals that can be used to feed additional power amplifiers connected to sub-bass bins and tweeter arrays.



The sub-bass and tweeter filters are 24 dB per octave, Linkwitz-Riley slopes, and the subsonic filter is 24 dB per octave Butterworth at 30 Hz. All outputs are balanced line drivers with a maximum output level of +26 dB into 600 ohms; input is active differential, transformerless-balanced. In addition, an EQ loop in the sub-bass circuitry enables insertion of an outboard equalizer (such as the future RLA Q5000 rotary program equalizer) or any special effects equipment.

The X3000A has a suggested retail price of $550.

RLA INTERNATIONAL, LTD.

For additional information circle #117

TANNOY SR840 POWER AMPLIFIER

The SR840 is a 250W per channel stereo amplifier designed for use in high-level studio monitoring, professional sound reinforcement systems and high quality public address work. The new amp is described to be the

result of "no-compromise" design and manufacturing standards: before dispatch each completed unit receives a full test on every specified parameter.

Quoted technical specifications include max RMS output power at clip point (+0, -0.5 dB) with both channels driven into 8 ohms of 250W, 440W into 4 ohms, 645W into 20 ohms, and 860W bridged mono into 8 ohms; THD and noise of less than 0.05% at any power over band 20 Hz to 20 kHz (typically 0.01% at 200W into 8 ohms); intermodulation distortion less than 0.03% (50 Hz and 7 kHz, 4:1 -12 dB); Unweighted noise ref 250W into 8 ohms -105 dB (A-weighted 116 dB); full power bandwidth +0, -0.5 dB 15 Hz to 40 kHz; and crosstalk at reference rated output at 1 kHz better than -75 dB. Output indicators have a rise time of 10 milliseconds, a fall time of 1 second, and ±2% of indicated level.

TANNOY NORTH AMERICA, INC.

For additional information circle #118

FAIRLIGHT UNVEILS CMI SERIES III DIGITAL SYNTHESIZER

Technical features of the new Series III include:

- Digital sound quality using 16-bit, random-access waveform storage and converters;
- A total 16 audio output channels, each with their own 16-bit DAC, VCF, VCA and output connector;
- Ability to route any of 16 voices to any of the 16 output channels;
- Massive common Waveform RAM to allow multiple samples across the keyboard, different waveform samples per keyboard section selectable or blended from performance dynamics;
- Several methods of off-line synthesis, including full Fourier analysis and resynthesis, FM and comprehensive sound manipulation and sustain techniques;
- Up to seven Megawords of Waveform RAM may be installed, providing 140 seconds of 16-bit sound when sampled at 50 kHz. (This time can be split as needed between instruments.) The sample time is more than doubled if WRAM is used in 8-bit mode (at 44.1 kHz sample rate: 5.2 minutes). Sixteen and 8-bit waveform modes can be intermixed, and up to 120 independent sounds accessed for each "Voice" (e.g. multisampling over split keyboard);
- Stereo sampling to WRAM at 50 kHz, mono sampling to 100 kHz;
- Playback sample rate to 200 kHz;
- External control via MIDI/SMPTE and click-track;
- Control of external equipment via MIDI, SMPTE in/out and TTL clock signals;
- A Real-time Composer sequencer can control up to 80 instruments simultaneously — 16 internally, and up to 64 via Fairlight's 16-voice slave racks or the four MIDI output sockets; and

- "Mega-sampling" capability to and from 60 Mbyte hard disk.

The main physical changes from Series IIX are an expanded motherboard, to hold the extra waveform memories, a 500-watt, switch-mode power supply and a 150-watt analog supply. A small high-resolution X-Y graphics tablet has also been introduced, in place of the previous lightpen, providing greater accuracy, ability to draw on black areas of the screen, and elimination of operator fatigue.

FAIRLIGHT INSTRUMENTS

For additional information circle #119

TASCAM SERIES 200 SOUND REINFORCEMENT AND RECORDING MIXERS

The M-208 (eight inputs) and M-216 (16 inputs) rack-mountable mixers feature four program busses, each with its own pan and fader, and a stereo bus also with separate faders. The program bus faders allow multiple input-channel signals to be grouped on a single fader for quick and accurate control during live performances, or recording.

The Series 200 also provides eight tape inputs with corresponding switches in the channel strips, so that an eight-track recorder can be left connected and brought into the mixer for playback monitoring.

Each channel strip provides: pad, trim, channel insert, overload indicator, two tape-selector switches, three-band shelving and sweep-type parametric EQ, effect and fold-

For additional information circle #120

back level controls, bus design switches, and pre-fader listen (solo) switch. Balanced and unbalanced inputs are available for all input channels, as are balanced and unbalanced outputs for the program busses and stereo bus. Sub-buss inputs allow the Mixer to accept signals from another mixer, to increase the available signal sources.

The Master Section includes, in addition to the program faders, pan controls and the stereo faders; effect return level and pan controls; master level controls for effect and foldback mixes; solo master level; and switching matrixes for the monitor and meters.

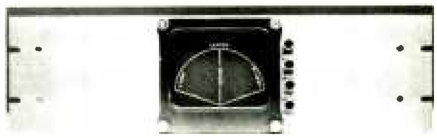
The M 208 carries a suggested retail price of $995, while the M 216 is priced at $1,495.

TASCAM

For additional information circle #123

IMAGESCOPE STEREO PHASE DISPLAY UNIT FROM B&B SYSTEMS

The new unit displays the complex stereo audio signal graphically showing the actual dispersion pattern of the sound energy as it will appear in a typical listening environment. The Imagescope provides a true visual realtime representation of the balance, separation and level of the stereo signal, and is intended for use in recording studio and audio production post-production facilities, in addition to the TV and AM broadcaster looking to stereo.



In recording and production, Imagescope is used to position exactly any track or tracks in the stereo image, in real-time, thus avoiding stereo phase errors. The unit is also used at transmission sites to verify separation and or mono compatibility.

B&B SYSTEMS, INC.

For additional information circle #124

SENNHEISER MKH-40-P48 STUDIO CONDENSER MICROPHONE

The combination of symmetrical capsule with optimal resistive loading is said to result in a highly linear frequency response, and in inherent noise level that is virtually imper-

ceiveable by modern digital recording equipment. The MKH 40 is described as handling both high and low sound-pressure levels with ease and accuracy, and is capable of recording the most subtle sonic nuance to the loudest boom without coloration, noise, or intermodulation distortion.

Featuring a cardiod directional pattern, the MKH-40 has a quoted frequency response of 40 Hz to 20 kHz, and a sensitivity of 25 mV/Pa (8 mV/Pa), ±1 dB. Nominal source impedance is 150 ohms, balanced, and minimum load impedance 1 kohm. Equivalent RMS SPL (DIN 45500) is 12 dBa (16 dBa), and equivalent peak SPL (CCIR 468) 21 dBa (26 dBa). Maximum SPL for less than 0.5% THD at 1kHz is quoted 134 dB (142 dB).

SENNHEISER ELECTRONIC CORPORATION

For additional information circle #125

CS-1200 POWER AMPLIFIER FROM PEAVEY

Peavey begins shipment of a new "power house" amp, the CS™ 1200. This unit is larger than their extremely popular CS-400™ and CS-800™ models that have been acclaimed as the industry standard in power amplification.

The new CS-1200 is described as a "brute" of an amplifier, built for ruggedness, reliability and superior performance. Large continuous duty power transformers are of a new semitoroidal-type, and the entire unit is designed to meet rigid European electronic and heat dissipation specifications.

The power amp has two separate channels, and its two massive power supplies are complemented by a large number of rugged silicon output transistors in a unique "tunnel-like" dissipation tube that maximizes heat dissipation: it is rated 600 watts RMS continuous per channel into 4 ohms, and 1,200 into 8 ohms in bridge mode.

As well as Peavey's patented DDT compression, the CS-1200 features the company's new "back porch" accessory plug-in patch panel for balanced input transformers, and crossover/special function modules.

Suggested retail price of the CS-1200 is $1,199.50.

PEAVEY ELECTRONICS

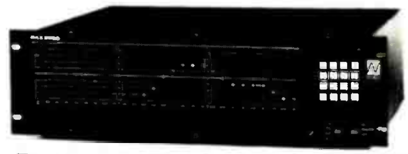
For additional information circle #126

NEI LAUNCHES DAX 2800 THIRD-OCTAVE ANALYZER/EQUALIZER

The DAX 2800 combines the functions of a third-octave, real-time analyzer with a digitally controlled third-octave graphic equalizer, and adds a computer for measurement and control functions. Under the control of the on-board computer, gain settings on each of 28 ISO-centered filtering bands can be stored and instantly recalled from 16 memory locations; specific requirements for EQ can be recalled on demand by the push of a button.

Additionally, the unit incorporates a digitally controlled RTA that can display the spectrum of an audio source on the 28-band LED matrix to locate room acoustic anomolies. Through the use of an internal pink-noise generator and sensing microphone, the DAX 2800 will perform computer-controlled automatic equalization and quickly EQ a system to any memory setting.

The unit also provides instantaneous, automatic EQ and RTA curve averaging of up to eight curves, and can be weighted as much as 7:1. Sensitivity of EQ or RTA settings can be changed from 3 dB to 1 dB.



For the sound contractor, the DAX 2800 acts as the "master" computer to download into the DAX EQ POD any prescribed room equalization curve, the latter unit is a blank-panelled, third-octave equalizer for permanent sound system installations, and cannot be adjusted by unauthorized personnel without the DAX 2800.

NEPTUNE ELECTRONICS, INC.

For additional information circle #127

NEW UREI MODEL 809 TIME ALIGN MONITOR FOR SMALLER CONTROL ROOMS

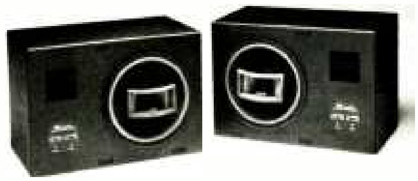
The 809 Studio Monitor features an all-new 12-inch coaxial driver that is said to deliver a true, one-point sound source, tight bass and superior stereo imaging. The new coaxial driver incorporates a titanium diaphragm compression driver in the HF section that

For additional information circle #128

New Products

extends response envelope to beyond 17.5 kHz.

When high- and low-frequency portions of sound do not reach the ear at the same time, the result is "time smear" which, UREI explains, can be fatiguing after hours of critical listening. The Time Align" technique utilized by the 809 solves "time smear" by considering driver placement and adjusting crossover group delay parameters to achieve simultaneous arrival of sound from the voice coils of the two transducers.



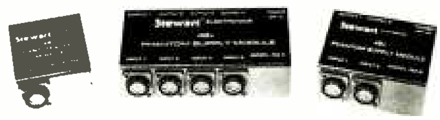
The unit utilizes the patented UREI HF horn with its diffraction buffer for correct acoustic impedance matching and smooth out-of-band response. Shadow slots eliminate the midrange shadowing common to conventional coaxial loudspeaker horns, while ceramic magnet structures are said to ensure that the system's sensitivity will not degrade with time or continual use.

JBL PROFESSIONAL

For additional information circle #131

STEWART ELECTRONICS EXPANDABLE PHANTOM POWER SUPPLY SYSTEM

The new expandable system is designed to be used with most microphones and accessories requiring 48 VDC.



Starting with the basic single-channel supply, the system is expandable to a full 12 channels, features full 48 VDC supply. Because of individual channel regulation, each channel is said to feature very low crosstalk, hum and noise. Low hum, coupled with isolation from input to output, permits the option of unbalanced operation, as well as providing protection for the inputs of equipment that does not have its own isolation.

The system also features short-circuit protection, which allows adjacent channels to remain unaffected in the event of a short.

STEWART ELECTRONICS

For additional information circle #132

SIMON SYSTEMS RDB-400 FOUR-WAY DI BOX

The new unit is a four-channel version of the DB-1A DI with added features and capability. The AC-powered RDB-400 is based on a totally active circuit design — there are no transformers in the audio path.

A three-position output level switch provides a "normal" output which, unlike other

— ADVERTISEMENT —

LAKE EXPANDS CAPABILITIES

What's new at LAKE? Besides the influx of new people . . . A host of new computer systems. Computers that assist in the design, engineering, drafting, and service of audio/video systems. One of the most exciting new computer systems is the audio departments Tecron TEF System 10. A portable audio spectrum analyzer that can be used in the field and the data brought back to the office for further analysis.

LAKE is involved in the design and building of television stations, recording studios, post production editing systems, and sound reinforcement systems worldwide. A computer system that could quickly analyze the acoustic parameters of any space was very important to the engineering department. They are currently using the TEF 10 to help expedite the engineering requirements of an expanding customer base.

An example of its value was recently discussed at a meeting I attended. It seems that microphones placed at a specific area on stage were experiencing excessive feedback. The client had tried a number of corrective measures to no avail. LAKE's engineers, using the TEF 10 were able to pinpoint the problem, something that at first glance seemed insignificant, a steam pipe located near the speaker cluster was causing a strong reflection into the problem area. Covering the pipe with absorbent material, eliminated the problem.

LAKE'S audio systems engineers Dennis Smyers (foreground) and Steve Blake analyze data on the TEF System 10

Without a doubt, this type of commitment on the part of LAKE in R & D, positions them as the systems company of choice in the audio field. Contact them at (617) 244-6881.

For additional information circle #130

DIs, has no insertion loss; "attenuation" mode with variable attenuation trim; and "line level" output mode with line trim for direct connection to a line input or tape deck, bypassing the console completely.

Because the unit is independently powered, design techniques are said to have been utilized that are not possible with a phantom-powered DI.

The RDB-400 can be floor-or rack-mounted, and has front and rear XLR, balanced outputs, and front-panel unbalanced buffered outputs. Suggested retail price of the RDB-400 is $895.

SIMON SYSTEMS

For additional information circle #133

RANE MODEL GE-30 THIRD-OCTAVE GRAPHIC EQUALIZER

Every aspect of the GE-30's design is said to have been fully optimized by computer analysis for maximum accuracy, minimum phase shift, minimum noise contribution, the

greatest flexibility, and total freedom from imprecise filter bandwidth variations common to other brands of graphic equalizers. Having been subjected to a thorough series of worst-case calculations by computer automation, the GE-30 is described as providing the highest levels of reliability and freedom from environmental stresses associated with the sometimes abusive requirements of commercial-audio installations.

Features include 30-band second-generation, state-variable constant-bandwidth filters; boost/cut (+12 dB boost and -15 dB cut) or cut-only (-18 dB) capability with LED indicator; switchable, active-balanced or transformer-balanced output; sweepable ultrasonic and sweepable subsonic filters; plus overload indicator and automatic "fail-safe" hard wire bypass with LED indicator.

RANE CORPORATION

For additional information circle #134

MODEL 310 COMPRESSOR/LIMITER/LEVELER FROM ROCKTRON

The Model 310 is a fully automatic compressor/leveler that offers program-dependent ratio, attack and release with selectable compression or leveling modes. The unit employs Rocktron's unique logarithmic compression for smooth compression, the company says.

The input gain switch allows the musician to plug his instrument directly into the Model 310 for use in live applications; when used in this mode, the musician may also use the 310 as a pre-amp.

The compression control simultaneously adjusts the threshold of compression and the input level to the compression circuit. Compression ratio automatically changes from 1:1 (below threshold) to 20:1, de on the program. Output level is adjus over a 25 dB range.

Two different modes of operation are selectable via the front panel: compression and leveling. In leveling mode, attack and release times are slow enough to have a minimal effect on program transients and short-term changes in dynamics.

The Model 310 also features a 10-segment gain reduction meter for monitoring the amount of gain reduction taking place.

ROCKTRON CORP.

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TANNOY FSM TWIN 15-INCH STUDIO MONITOR

The new FSM is an addition to the company's dual-concentric SRM Series of monitors. Improvements in specifications are said to have resulted in a high power, high sensitivity, studio monitor capable of resolving fine detail at extremely high SPL.

The cabinet constructed of 1.25-inch medite (a very dense, hard, manufactured timber product) is divided into two totally separate internal chambers. The bass reflex system employs one 15-inch dual-concentric for mid/high frequencies, and one 15-inch transducer for bass.

The system also utilizes an all-new Low Frequency Window for bass equalization and Hard Wire crossover technology.

Suggested Retail of the FSM is $4,198.

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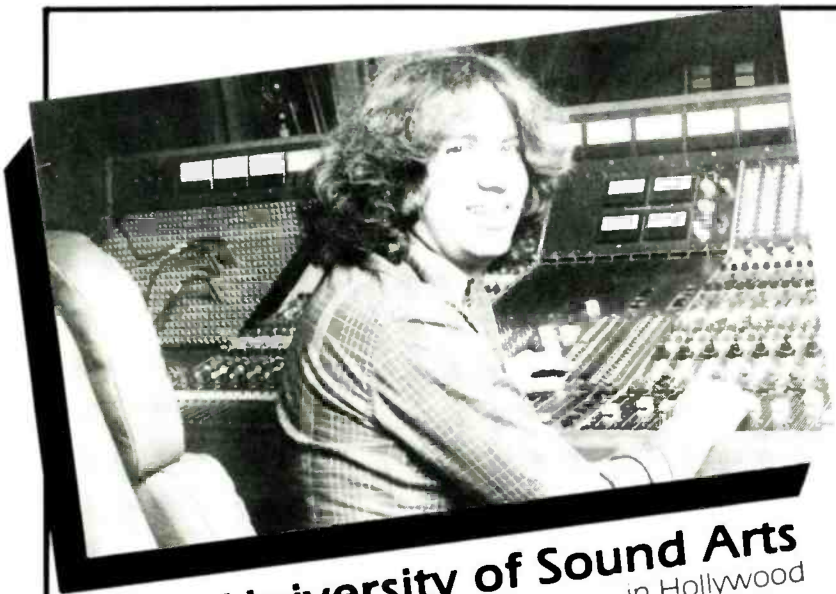

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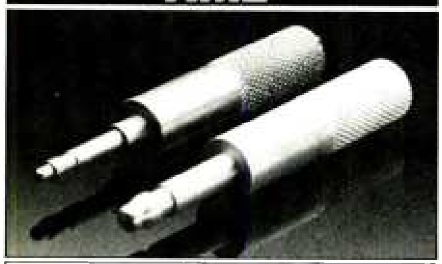

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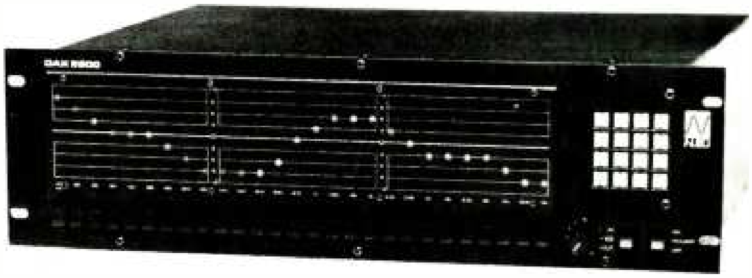

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News

— continued from page 26 . . .

special projects, will handle the production of live concerts on the East Coast, utilizing his own custom-designed mobile recording facility. Starfleet's mobile will join Westwood One mobile studios based in Los Angeles and Cleveland. Most recently, Starfleet produced a worldwide radio broadcast of the *Live Aid* concerts from Philadelphia's J.F.K. Stadium.

• **PEAVEY ELECTRONICS CORPORATION** recently was presented with the Presidential "E Star" Award for continued excellence in export achievements. In accepting the award at a special ceremony in Meridian, Mississippi, company founder and president Hartley Peavey said: "I want to personally thank all of our loyal distributors in the 92 countries around the world for making this possible. Without their support, we would not be able to accomplish this major goal." Eligibility for the award in exporting requires that a company be selected previously for the Presidential "E" Award, which the company received in 1978.

• **EMI MUSIC** has announced plans to build a Compact Disc mastering and manufacturing facility in the U.K., to be located at Swindon, Wiltshire. The new plant is scheduled to begin initial production early next year, with annual capacity gradually increasing to between eight and 10 million CDs. The new capability will significantly increase the company's Compact Disc supplies, which currently are being met by manufacturing services in Europe and Japan. At present, Nimbus Records is reported to be the only company to be manufacturing Compact Discs in the U.K.

• According to **GOTHAM AUDIO** service laboratory manager, Joe Leung, parts for restoring **NEUMANN U47 and U48** microphones are now available from the company. Metal parts for U47/48 mikes have been unavailable since the early Seventies, Leung explains, but now Neumann has begun re-manufacture of a limited supply of housing tubes, head grilles, output transformers and other items. Further details of restoration service costs are available from Leung at: (212) 741-7411.

• **THE PLANT STUDIOS**, Sausalito, CA, has introduced a mobile maintenance service for the San Francisco Bay Area. Complete with mobile van, radio dispatch, and utilizing a Sound Technology Model 1510 Tape Recorder Audio test System, The Plant's maintenance team will be on-call 24 hours a day to service recording and production studios. As part of the new service, the facility also is offering a Preventative Maintenance Program that will provide checkups of studio equipment on a regular basis. "Maintenance is a mutual

. . . continued overleaf —

News

problem that all recording studios share in this industry," says Plant owner Stan Jacox. "Now it's time, in the Bay Area, to make it a mutual asset." More details of the new service are available from The Plant Services: (415) 332-6100.

• **PROFESSIONAL AUDIO SERVICES** has relocated to larger premises in Burbank, CA. According to owner Bob Hacken, the company is now based in the former site of Kendun Recorders, whose 5,000-square-foot facility features an "acoustically-correct" demonstration room measuring in excess of 500 square feet. The address of the new location is 619 South Glenwood Place, Burbank, CA 91506; the company's telephone number remains (818) 843-6320.

• **WOLFF ASSOCIATES** has purchased the entire API product line of recording and production consoles, plus modular signal processors. The final transfer of ownership, which took place in late June, means that the company will now be manufacturing the API console line once again, as well as developing a new range of products that is expected to include retrofits for older consoles and a moving-fader system. To date, one authorized dealer for the new product line has been appointed: **Studio Consultants**, New York; a network of distributors will be chosen within the next few months, the company says.

• **OCEAN AUDIO**, the Los Angeles-based used pro-audio equipment dealer, has rented a Solid State Logic SL4000E console to **AMIGO STUDIOS**, North Hollywood. The SSL board previously was in service at Ridge Farm Studios, England, and features a 40-input mainframe with full automation capabilities. According to Ocean Audio president David Hadler, Amigo is renting the console on a three-month renewable lease, at a rate of 6% of the SL4000's resale value per month; the studio also has the option of purchasing the board at the end of each lease period. The Amigo lease is Ocean Audio's first in the console rental market, and may indeed represent the first such rental deal of a Solid State Logic board in the U.S. "There's a substantial number of studios that would like to seriously work with the SSL consoles," Hadler offers, "but not all can afford the luxury of placing an order for a six-figure piece of hardware. We've put together a very reasonable and affordable rental/purchase program for high-end consoles that puts them within easy grasp of just about anyone who's serious about the kind of business boards like these could generate for their studios." The company also is reported to be arranging for the delivery of several additional SL6000 Series consoles for the U.S. rental market. More information is available from David Hadler on (213) 454-6043.

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STOP PRESS:
Mitsubishi Pro Audio Group Established in UK

Digital Entertainment Corporation, a subsidiary of Mitsubishi Electric Sales America, Inc., has announced the start of full sales and service operations in the United Kingdom, under its recently acquired subsidiary, Quad-Eight/Westrex, Ltd. Formation of Digital Entertainment Corporation, Ltd., is currently in process, which company will act as the overall UK subsidiary headquarters. The primary trading name to be used in the UK is Mitsubishi Pro Audio Group, which means that both the North American and the UK operations will be integrated under one common control.

Peter Sidey has been appointed special executive consultant with full responsibilities for Mitsubishi Pro Audio Group UK operations; he previously served as managing director with Neve during the Seventies.

Adrian Bailey has been appointed manager of Pro Audio Marketing, Manufacturing and Technical Services at the UK headquarters; he joins the company with 15 years experience at Neve Electronics.

Barry Motton will continue in his position as managing director of Quad Eight/Westrex, Ltd.

"We have been working a long time to get to this point," says Mitsubishi Pro Audio Group president Tore Nordahl. "With the resources of our $8 billion-strong Mitsubishi Electric parent company, our commitments to the UK market are substantial and long term. We are putting our energies into key pro-audio products in the major markets around the world. Our three-year plan is to be the leader in digital recording, film recording, and in digital/analog consoles."

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The Wheel Thing. The A820's dual thumbwheel shuttle/edit control makes tape-cut editing a breeze. One wheel fast winds tape in either direction at increasing speeds while the other precisely positions tape for the edit.

Also Noteworthy. The A820 incorporates Studer's new generation of phase compensated audio electronics, available with either transformer or active balanced inputs and outputs. In sound quality, the A820 takes a quantum leap ahead of recorders made just a few years ago. Options for the A820 include a center-track SMPTE time code channel and test generator.

The Payback. The "hardware" in the Studer A820 is made to give you dependable service for years to come. That's the Studer tradition. Plus, with its advanced software, the A820 also does more different jobs, gets more jobs done in less time, and produces sonically superior results. That's the bottom line in any upgrading program.

For more information on the new A820 Analog Master Recorder, please contact: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

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