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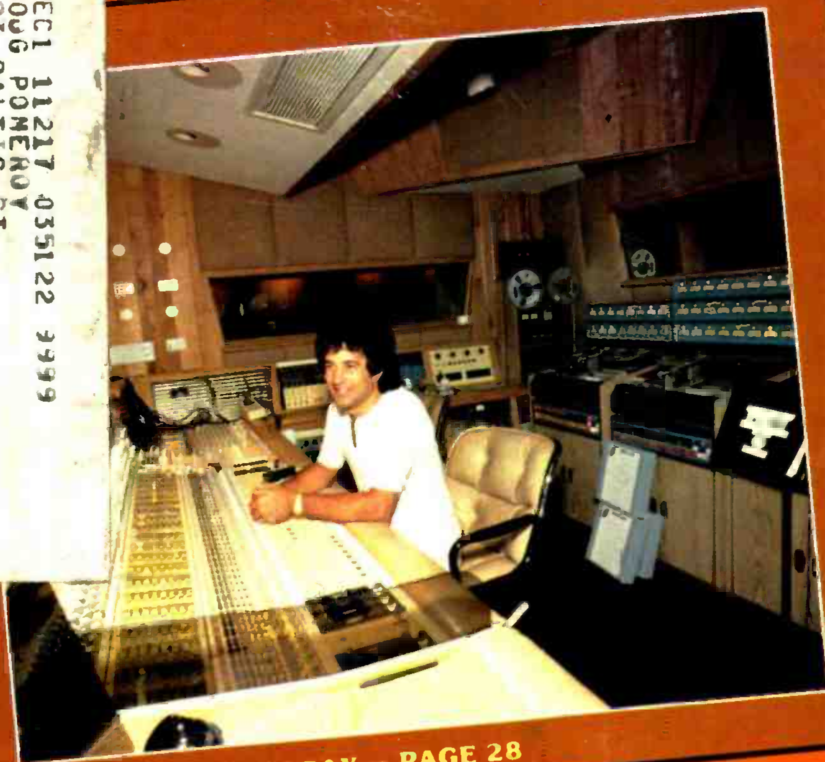
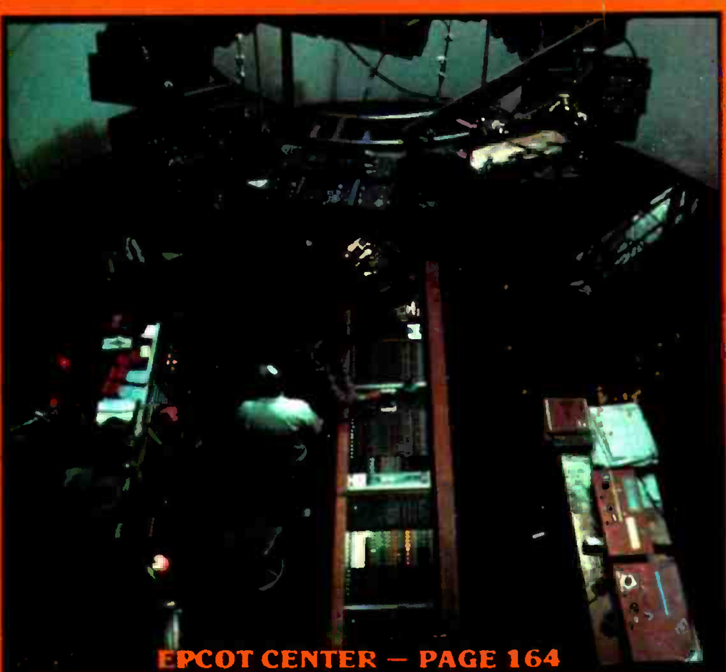
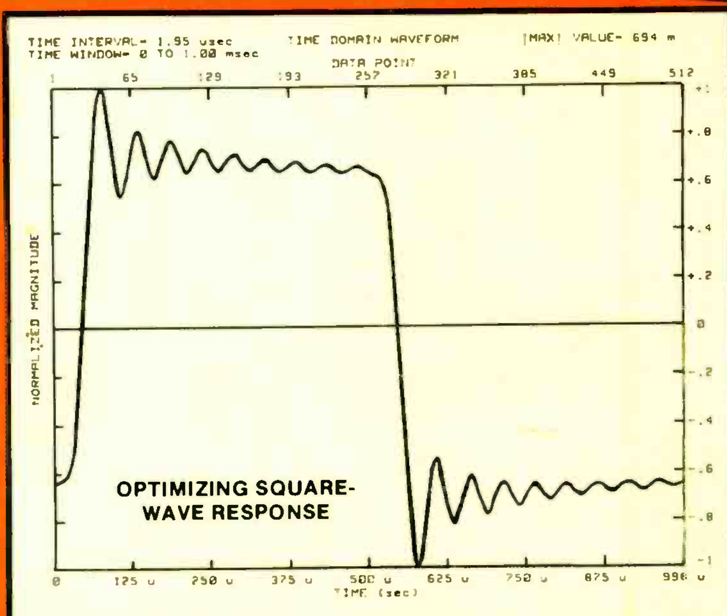


PRODUCER VAL GARAY — PAGE 28

AUDIO FO



EPCOT CENTER — PAGE 164



CIRCUIT DESIGN — PAGE 174

SOUN

US FESTIVAL — PAGE 94

R VIDEO — PAGE 54

STUDIO DESIGN — PAGE 126

M SOUND — PAGE 150

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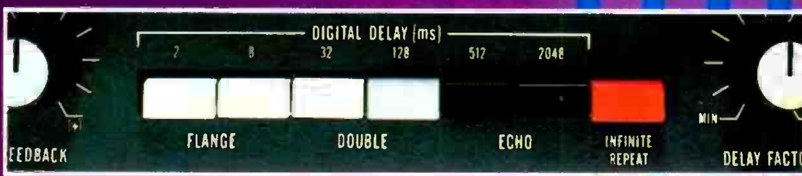
DELTALAB'S NEW *SUPER* TIME LINE



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If you're banking on professional audio for your future success, you'd better make a commitment to digital audio. It's clear that digital audio is the future of entertainment audio production. As a U.S. affiliate of Mitsubishi Electric Corp., Digital Entertainment Corporation has the technological and financial resources to make sense of your digital audio investments. Your first important step is to evaluate the equipment and systems available. We're certain you will find the X-80 to be superior from all points of view. Just check the following features:

X-80 IS AFFORDABLE

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

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

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

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The X-80 reads after write, allowing continuous monitoring of what has actually been recorded. Try that with the video cassette based systems.

ANALOG CUEING TRACK

The X-80's additional analog cue track allows you to listen to the tape at any tape speed for editing and cueing purposes. (Remember that digital tracks can only be decoded and monitored at nominal tape speed.) Try that with the video cassette based systems.

SMPTE/ADDRESS TRACK

A separate address track is provided for easy interface with other audio and video systems under synchronizer control, without "stealing" one of the audio tracks.

DIGITAL TRANSFER AND COPY FACILITIES

X-80 digital-to-digital copy and transfer facilities are available in New York, Los Angeles and San Francisco right now. Nashville and Toronto will follow shortly. Standard converters will be available for all significant formats by the end of 1983 to assure full compatibility with compact digital audio disc master requirements, Also, we would be delighted to assist in the transfer of "old" digital audio libraries to the X-80 standard.

THE RIGHT DEAL FOR YOU

We want to help you prepare your future in digital audio, whether it is stereo master or 32-channel multitrack recording, or electronic editing systems. As a member of the Mitsubishi Electric Corporation ($6 billion world-wide sales in 1982), our commitment to you is substantial and meaningful. Discuss your plans with us before you make an investment in analog or digital audio tape storage or console systems. Invest your dollars carefully, especially in analog. Call Bill Van Doren, Lou Dollenger or Tore Nordahl. You're assured straight answers, fair dealings and long time audio production equipment expertise. And we'll do our best to tailor the right equipment and financial package to your situation.

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DIGITAL AUDIO SYSTEMS

For additional information circle #2

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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October 1983
Volume 14 — Number 5

— Contents —

— Departments —

Making the Best Even Better

New England Digital Announces Exciting New Synclavier® II Options

Roland GR Guitar

New England Digital Corporation is pleased to announce exciting new options to be available as additions to all Synclavier II's.

Once again, with the release of these options, New England Digital honors its commitment to steadily upgrade the Synclavier II.

This consistency could only be possible from a company whose product incorporates tremendous hardware/software flexibility and power. ***The Synclavier II has them both!***

DIGITAL GUITAR OPTION

Guitar Players - welcome to the world of computer music! The synthesis power, creativity and flexibility of the Synclavier II can now be offered to guitarists using New England Digital's revolutionary new digital conversion process. Complete access to all of Synclavier II's capabilities, such as digital synthesis, automated music printing and Sample-to-Disk™ are now yours. A unique 16-button LED panel attaches simply to all Roland GR* guitars to allow convenient access to important Synclavier II real-time features. (Available August 1983)

Optional Ebony Model Synclavier II Keyboard

* Roland GR is a registered trademark of Roland Corporation, Japan

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STEREO OPTION

Now any Synclavier II can be simply upgraded to produce fantastic live stereo results. Many elaborate stereo control modes never before possible from any system or recording environment come standard with Synclavier II's new Stereo Option. Increase your Synclavier II's sonic capabilities, *plus* save valuable production time and expense by going direct from Synclavier II's 16-track digital recorder to 2-track tape!

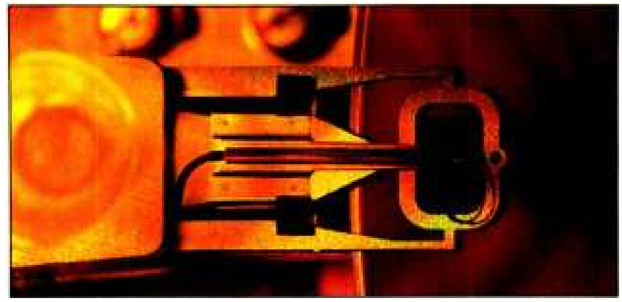
MUSIC PRINTING "Enhanced"

In March of 1983, New England Digital released a new, enhanced version of software for Synclavier II's Music Printing Option. Now, important aspects of western music notation such as random changes in time signatures and key areas are available, along with tuplets of any kind. Plus, you will have instantly accessible editing capabilities along with dynamic markings to enhance your finished complete score or individual parts (see example below). Yes, there is an automated commercial music printing system which is available today . . . *and works*.

Actual Music Printing Sample, Reduced

SAMPLE-TO-DISK "Polyphony"

The company which offered the only high fidelity sampling system worldwide with a sample rate of *50kHz, 16-bit data conversion*, and extended sampling time to Winchester Disk (pictured below) is planning an exciting new enhancement for the Synclavier II's Sample-to-Disk option . . . POLYPHONY. New England Digital engineers are now working to expand the sampling capability to be completely polyphonic. The same high-fidelity sonic capability and high resolution presently offered will be incorporated. The new polyphony option promises to add one more amazing capability to the Synclavier II.



Z-80/C.P.M. OPTION "Personal Computing"

Available for all Synclavier II systems is the convenient Z-80/C.P.M. option. This simple retrofit option allows all users to purchase computer industry standard C.P.M. software programs to aid their personal or company computing needs. Whether it is accounting, word processing, or computer games, New England Digital's Z-80/C.P.M. adds another dimension to the remarkable Synclavier II.

To New England Digital these additional options are just steps along the path to the ultimate instrument. Some day we believe the Synclavier II will be a complete music production facility. We also know that it takes a series of developments to achieve this goal, especially in this highly technical field. Using New England Digital's advanced hardware and software, along with creative input from customers, will ensure the longevity of the Synclavier II. We invite you to start with the best and grow from there, as hundreds of others have! If you haven't heard a Synclavier II lately, you haven't heard it at all!

Synclavier II Instruction Manual
A complete and descriptive Instruction Manual is available for $85 (USA & Canada) and $100 US (elsewhere).

For more information please call or write:

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views letters news

FOUR-TRACK RECORDING TIPS

from: **Dan Dugan**
Dan Dugan Sound Design
San Francisco, CA

I'd like to comment on a couple of points raised in Bruce Black's fine article, "Achieving Best Results from Eight-Track/Half-Inch"; *R-e/p*, April 1983 issue. The following are lessons I've learned from doing professional production and service work on quarter-inch, four-track machines, and are referenced to some of Bruce Blacks' comments:

1. "Sometimes a recording recorded on one dbx unit will not decode properly on another unit, resulting in the signal level 'pumping' in time with a kick or snare drum, or some other percussive or bass-heavy instrument." I have experienced this too, but I don't think it has anything to do with different dbx units. This kind of mistracking is caused by one or both of the following two effects:

a) Attempting to record frequencies lower than the response of the tape deck. The bass frequencies are part of the dbx record encoding, but the recorder is an effective sharp 40-Hz high-pass filter. When the sound is played back those frequencies aren't in the mix anymore, and the decoder will mistrack. The solution is to use a 40-Hz high-pass filter between the console and the dbx record encoder.

b) Almost all recorders have significant "head bump" frequency-response errors at the low-end, typically 1.5 to 3.5 dB. These built-in errors are audible as a "thickening" of the bass, and also cause noise reduction to mistrack or pump. The solution to this is an equalizer carefully adjusted to flatten the frequency response of the deck, patched between the deck's play output and the input of the dbx decode unit. (I developed an equalizer for this purpose which I manufacture in small quantities for those few who can appreciate the difference.)

2. "A phenomenon known as 'fringing' causes the high-frequency response of edge tracks to noticeably fluctuate or 'crackle.' " This terminology is incorrect. "Fringing" is an effect seen when playing a full-track test tape on multitrack heads. The bass frequencies play back at higher levels because the head reads an area wider than the actual head core at low frequencies.

Everyone who works with narrow tracks knows "edge-track flakeyness." The good news is that it can be almost completely eliminated by: 1) using only the best quality tape; 2) careful alignment of heads at the time of installation; 3) careful handling of tapes; and 4) undecutting the heads so that no wear step develops at the edge of the tape.

In my shop it is standard practice to send all new heads to Taber (2468 Embarcadero Way, Palo Alto, CA 94303) for their "edgecut and deburr" service, which costs $25 per four-track head. With properly installed undercut heads the edge tracks will stay solid forever.

I should mention that some engineers are in the habit of installing four-track heads too low on purpose, so that three tracks will be good and the bottom track extra-flakey. Undercutting makes this practice unnecessary. Try it, you'll like it!

CONNECTOR POLARITY

from: **Dennis A. Bohn**
VP Engineering,
Rane Corporation
Mountlake Terrace, WA

I have been thoroughly enjoying John Roberts' series entitled, "Exposing Audio Mythology." John is covering many of my favorite myths. There are so many that I can imagine his series could be a very long running one!

Regarding John's call for what manufacturers are doing polarity and 3-pin connector-wise: *R-e/p* readers might like to send for a free copy of our Rane Note 102, which covers just these subjects. We, too, feel that connector polarity is an extremely important and badly neglected area of product design by the entire professional audio industry, and are doing our part in trying to get the message out to anyone that is willing to listen.

There is only *one* standard for 3-pin connectors: IEC 268, part 12, "Circular Connectors for Broadcast and Similar Use," 1975. Signers were Australia, Belgium, Canada, Denmark, Egypt, Hungary, Israel, Japan, Netherlands, Norway, Romania, South Africa, Sweden, Switzerland, Turkey, UK, and USA. Rane wires its 3- and 5-pin connectors per IEC 268 as follows:

For 3-pin connectors — Pin #1 *ground* (shield, screen, etc.); Pin #2 *positive* (signal, hot, etc.); and Pin #3 *negative* (return, common, etc.).

For 5-pin connectors — Pin #1 *ground;* Pin #2 *left-positive*; Pin #3 *left-negative*; Pin #4 *right-positive*; and Pin #5 *right-negative.*

I do encourage John Roberts to follow up with the list of what manufacturers are wiring to what standards. This will only encourage everyone to adapt to the standard that the U.S. has signed.

Thanks again, for being a rational voice amidst the jungle of hysteria out there!

From the Publisher:

It is with great pleasure that we announce the promotion of MEL LAMBERT to Editor of *R-e/p*.

Mel joined *R-e/p* initially as Editor-at-Large in 1980, and became Managing Editor concurrent with his U.S. residence status in 1983. Mel's appointment as the Editor of *R-e/p* is consistent with, and recognition of his superb understanding of, the workings and needs of the many facets of the professional audio industry, as well as the contribution he has made to the continuing progress and authority of the editorial product appearing in the magazine.

Mel will be ably assisted by a team of consultant editors, all of whom are highly regarded in their respective fields of activity, and in the day-to-day operation of the magazine by Assistant Editor, Sandy St. Claire.

AUDIO MYTHOLOGY: SPECIAL EFFECTS

from: **Marvin Caesar, president**
Aphex Systems, Ltd.
North Hollywood, CA

I would like to thank John Roberts for his reference [*R-e/p* August issue; "Exposing Audio Mythology"] to Curt Knoppel's patent, which is the basis for the various models of the Aphex Aural Exciters. I would like to point out that Aphex Systems, Ltd. has exclusive use of this patent, and holds a registered trademark on "Aural Exciter."

Manufacturers of devices which are labeled "exciters" purposely do not use any harmonic generation in order to avoid any patent infringement. Therefore, it is misleading to *R-e/p* readers to use the generic term "exciters" for all the devices, and describe the functions and methods which are exclusive to the Aphex Aural Exciter units.

By the way, the patent system was set up with the intention of advancing technology. But it would not work unless inventors did enjoy the monetary benefits of having patent protection.

SPECIALIZATION: WORKING DIRECT-TO-TWO-TRACK

from: **Jeffrey Weber**
En Pointe Productions
Beverly Hills, CA

I was intrigued by the premise of James Riordan's column in the June 1983 issue of *R-e/p*, concerning diversification and specialization. I am an independent audiophile producer and, to my knowledge, there are no other strictly-audiophile producers working

— continued on page 16 . . .

THE MCI PROMISE.

When MCI engineers set out to design the world's finest recording/remixing console, they promised to pack every one with more performance and more features than ever before. And they kept their promise. So well, in fact, that MCI consoles have become the choice of leading recording studios.

Now Sony is making a promise. The same versatile MCI JH-600's that turn out hit after hit can tackle every new mixing and sweetening job you have, head-on. Audio production for broadcast, teleproduction, or wherever your studio is moving, one of Sony's four flexible MCI JH-600 consoles can move right with you. You have our word on it.

For further information contact:
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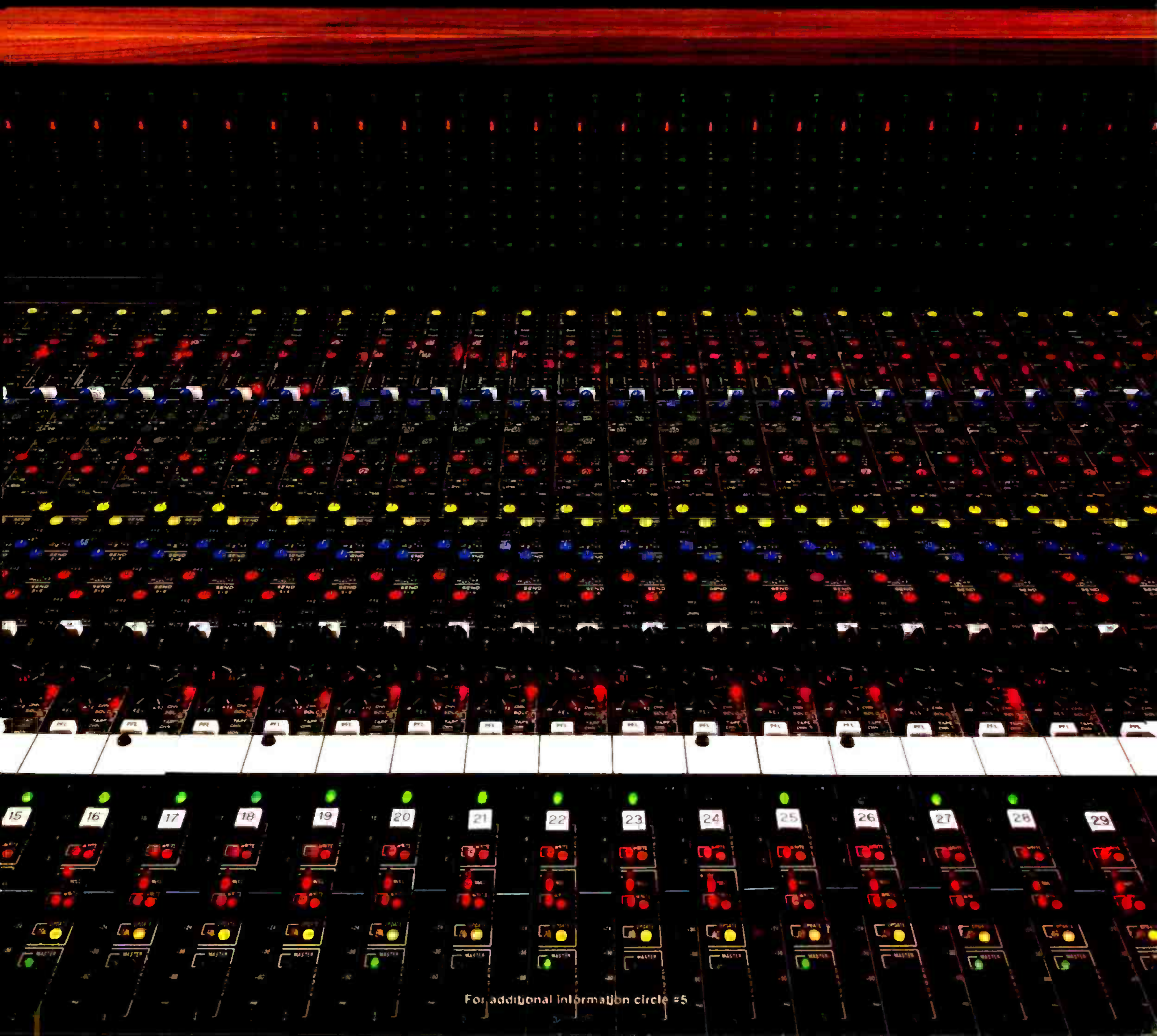

For additional information circle #5

HALLS
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SPLITS
BANK 5
CONCERT HALL
CHORUS & ECHO
B4 PROGRAM 1
DECAY

LETTERS

. . . continued from page 12 —

on their own. While I believe in, and have experience with, multitrack recording, I try to convince my artists that recording *live* to two tracks is to their best advantage. I explain that music should be the collective expression of creativity by a *group* of players.

Layering or stacking tracks, while definitely having certain supposed advantages, namely flexibility and error forgiveness, does not yield the emotional performances that I believe make for exceptional recordings. The time involved in creating a multitrack album turns the creation of musical expression into "work" — something to be tolerated and endured, but generally not looked forward to.

Recording live requires no more than two days in a studio; there is no mixing, editing, or overdubbing. If three keyboard parts are called for, three keyboard players are called to the session. Thus, we create an *event* by having all the players and singers together in the same room. We create an emotional surge that translates to disk very well. We want the listener to *feel* as well as *hear* the results.

Naturally, recording such live projects is not "safe." Everyone on both sides of the glass has to be totally prepared. While multitracking creativity often takes place in the studio, live two-track creativity occurs prior to entering the studio, either by rehearsal (perhaps in another, smaller studio), or by subjecting the material to audiences. Once the performers feel confident in their abilities, as well as with the material, the emotional surges of the moment often take their playing or singing to higher levels of virtuosity. The results often can be euphoric.

With suitable preparation, all types of music can and have been recorded in this fashion, including rock. As can be seen, there is only so much money you can spend in a two-day period. Studio time for my projects has never gone above $7,000 for the entire LP. We have been very successful in coming in very close to our budgets, and the albums end up sounding remarkably superior to most conventionally recorded projects. Naturally, I work with some of the most gifted engineers available — Allen Sides, Rik Pekkowen, Rick Ruggieri, Ron Hitchcock, Mark Ettel, to name but a few — and that alone is a large part of the product, since these engineers must be capable of recording and mixing *live*, a talent that is hard to come by these days.

When the parts are put together, you have an album that is completed in 95% less time than most other projects, generally sounds up to twice as good (if I do say so myself), and costs between 50 to 75% less than conventionally recorded albums.

NEWS *begins on page 213 —*

CONTINUING A TRADITION OF EXCELLENCE.
THE SSL SL4000E SERIES MASTER STUDIO SYSTEM ARRIVES AT ABBEY ROAD, STUDIO 2, SEPTEMBER 1983.

Solid State Logic

EXPOSING AUDIO MYTHOLOGY

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

by John Roberts

In past columns I have pointed out some of the problems hindering communication between "Golden Ears" and "Meter Readers," and we have started to compile a GE Glossary (keep those cards and letters coming!). Besides the use of rather imprecise terminology, the GE commentary has a tendency toward hyperbole (or exaggeration for effect). It becomes difficult to take an equipment reviewer seriously, when he (or she) rants about the astounding sonic differences between two brands of welding cable hooked up to a 10-foot speaker run. (More on wire later). Perhaps I'm also exaggerating (for effect!), but my point is that little effort is made, nor does a system exist, to quantify the magnitude of these *audible* differences.

A more rigorous analysis of why two products sound different would come up with hard numbers (I don't believe in magic). But we can't expect product reviewers to have the instrumentation and experience to evaluate the sonic consequence of *minute* differences. Often the sonic differences are so slight that it is a judgement call as to which sounds better. Sometimes they just sound slightly different. However, when two products sound different it is human nature to prefer one, sometimes even when they *don't* sound different. There are no ties in the land of high-end HiFi.

I realize that it may make for dull reading, but in the many cases where the sonic differences are slight let's call it like it is. If the mailman can hear it through the screen door then maybe you've got something.

Speaker Wire

When I sat down to write this piece, I was prepared to debunk all those fancy HiFi cables as pure hype. Well . . . just like any good myth, if you dig deep enough there will be a grain of truth, however tiny. The basic premise of the esoteric speaker-wire proponents is that common speaker cable can alter the sound quality of electrical signals passed through it. Now anyone who has ever talked over long-distance telephone lines can attest that there is a loss of sound quality when you travel over many miles of wire, but what about a few feet? Well, yes, there is a potential for signal degradation in runs of only tens of feet but, as is often the case, the audible significance is up for debate

Wire varies from ideal in several ways. Wire pairs have capacitance, inductance, skin effect, dispersion, and last, but not least, resistance. To optimize a speaker cable it is desirable to keep all of these non-ideal characteristics to a minimum. Capacitance should be kept low to avoid destabilizing the driving amp, or otherwise interacting with the amplifier's output impedance. Inductance should be kept low, because it will cause a high-frequency rolloff when terminated by a resistive or, worse yet, capacitive speaker load.

Skin effect is a rather unusual phenomenon, where at higher frequencies the electrons tend to travel on the *surface* of the wire. This effect can cause significant changes in resistance between DC and high audio frequencies, in some cases diminishing the advantages of larger gauge wire. At 20 kHz, for example, the electrons that cause current flow are only using the outside 0.5mm of copper wire.

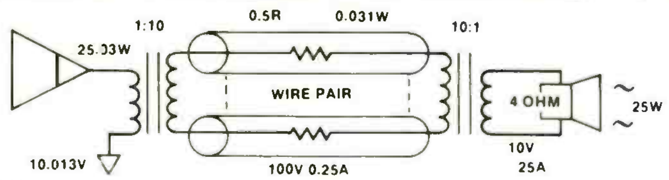


Figure 1

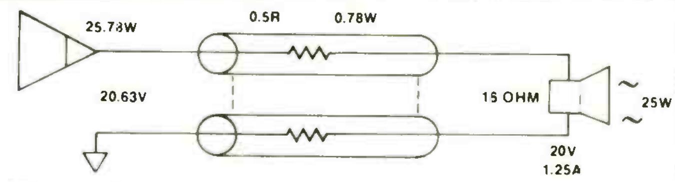


Figure 3

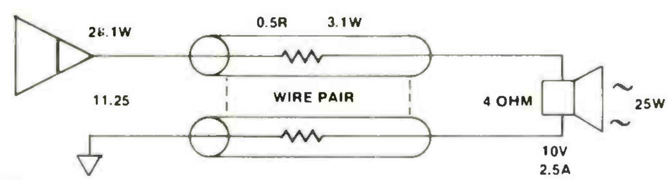


Figure 2

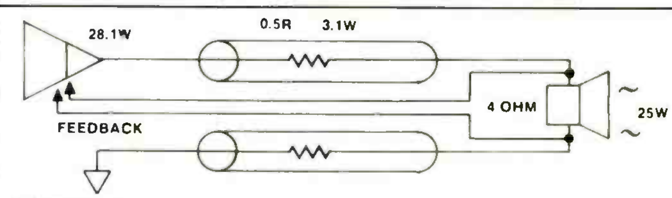


Figure 4

Dispersion is another interesting characteristic — in typical wire pairs the high frequencies travel slightly faster than the low frequencies. This was a major problem for long-distance telephone lines until the 'phone companies figured out what was happening, and added coils (inductance) to slow down the high-frequency content and time align the telephone signals.

Last, but *definitely* not least, is resistance. If speaker loads were a pure resistance then we could just live with the power drop in the wire. But speakers have a nasty habit of changing impedance with frequency which, if there is significant wire resistance, will cause a frequency-response error. Also the series resistance will degrade an amplifier's damping factor.

There are techniques to reduce some of these non-ideal characteristics, but usually they are made at the expense of another parameter. Braided wire will exhibit very low inductance, but often at a cost of higher capacitance (upwards of 1500 pF per meter). While spacing the leads further apart will reduce the capacitance, it also increases the inductance, as well as susceptibility to crosstalk. Going to very large gauge wire can reduce series resistance but, unless the wire is made up of many fine strands, the skin effect can cause large, frequency-dependent resistance changes. For example, 12-gauge wire can change resistance from 0.01 ohm per meter at DC, to 0.015 ohm per meter at 20 kHz. For the kinds of distances we are likely to encounter, dispersion is not a serious problem (usually less than a microsecond).

Is It Audible?

Recently, a controlled listening test with HiFi speakers and 30-foot cable runs demonstrated that the differences between 24- and 16-gauge speaker wire were clearly audible (refer to reference #2 for further details). This is not too surprising when you consider that the use of 30 feet of 24-gauge wire is like adding a 1.8-ohm resistor between the power amplifier and speaker. In this case, with a nominally 8-ohm HiFi speaker the variations in speaker impedance caused three discrete, -1.25 dB dips in the audible range. The same listening panel was unable to discern (with any statistical significance) differences between 16-gauge cable at 0.24 ohms, and a well known high-performance cable at 0.09 ohms driving the same speaker.

The implications for control-room monitoring, or home listening situations, is that speaker wire is *not* a serious problem. In live PA, or distributed systems, however, where long wire runs are used, the series resistance *can* be significant. Let's take as an example a nominally 4-ohm speaker system that varies from 3 to 12 ohms over the audio band. When driven by a wire having a 0.5-ohm series resistance, the system output will vary a full dB. A 1 dB response error can be quite audible, depending upon how smoothly the impedance of the speaker system varies. Even if the speaker was a dead flat 4 ohms, the wire would be dissipating *11%* of your power output.

To obtain a 0.5-ohm resistance, you would need: 8.3 feet of 24-gauge wire; 60 feet of 16-gauge wire; 150 feet of "Wazuu" speaker cable; or 1,600 feet of welding cable. (Note: at 20 kHz this latter cable will exhibit a resistance of 2 ohms, because of skin effects.)

What Does It All Mean?

Since the resistance and most of the non-ideal characteristics are a simple function of cable length, it is better to mount the amplifiers near the speakers, and keep the wires as short as possible. If you *must* drive a very long feed, it will be worth doing some homework. When looking at a new speaker purchase, ask the manufacturer for an impedance plot. If you are working on an existing system, simply measure the frequency response at the speaker terminals. (Note: be sure that the wire is played out; leaving it coiled up could affect the results.) You should be able to quickly evaluate the effect of changing to a lower resistance wire, by using a shorter length of your regular wire. If the response errors are smooth, it may be reasonable to try equalizing them out. However, in general it is always better to eliminate a frequency response problem, than to "fix it in the mix."

There are a few techniques that may lend themselves to the problems of non-ideal wire, which are worth investigating. The power loss in long wire runs is called "IR losses," and is defined by the equation, $P = I^2R$. So called "high-voltage" distribution systems reduce the power loss by reducing the current flowing through the wire, as shown in Figure 1.

By way of an example, suppose we are putting 25 watts into a 4-ohm speaker, with a 1:10 step-up transformer at the amplifier end of a 0.5-ohm wire feed, and a 10:1 step-down transformer at the speaker end. The voltage and current at the speaker is 10V and 2.5A respectively. Assuming a perfectly lossless transformer, the high-voltage loop will be passing 100V at 250 milliamps (still 25W), and the wire would drop 125 millivolts (0.25×0.5) for a dissipation of 31 milliwatts. Assuming another lossless transformer at the amp end, the power amp needs to put out 10.013V and 25.031 watts (yielding 99.9% efficiency).

Without the transformers in circuit (Figure 2), the wire would drop 1.25V and 3.1 watts, requiring the amplifier to put out 28.1W to drive the speaker to the same loudness (88.9% efficiency; 11.1% dissipated in the wire). Without debating the merit of using transformers in signal paths (that argument will be the topic of a subsequent column), typical PA power levels would require transformers so large that you would be better off just flying the amps (and keeping the cables short). Secondly, who needs another high-voltage source kicking around?

On a more sensible scale, however, if you are trying to decide whether to wire your PA for 16 or 4 ohms, and your power amps can put out the voltage swing, go for the higher impedance, since it will reduce cable losses and frequency-response errors. To return to the 25W example: following the circuit shown in Figure 3, the amplifier now would have to swing 20 volts into the 16-ohm load, but the same 0.5-ohm wire only drops 625 millivolts and 780 milliwatts for 97% efficiency.

Another approach worth looking at here is the use of "sense lines," a technique borrowed from laboratory power

supply designs, where a second wire pair is connected to the load (speaker leads) and returned back to the amp (Figure 4). Since there is no current flowing in these leads, the amp can use them for feedback and regulate the voltage at the speaker terminals. This approach can regain damping factor and frequency response, but will still suffer the IR losses in the cable. While such a design approach has shown up on one or more consumer amps (for those killer 15-foot speaker runs), I'm more than a little nervous about hanging sense lines on a load 200 feet away. With proper shielding and care, however, it might work.

In my opinion the only good speaker lead is a *short* one.

Reading For Extra Credit:

Reference #1 provides a rigorous analysis of a speaker wire's non-ideal properties. Reference #2 documents the double-blind listening test between speaker cables mentioned above.

1. R. A. Greiner, *Amplifier-Loudspeaker Interfacing:* Journal of the Audio Engineering Society; Vol. 28; pp. 310-315 (May 1980).

2. L. Greenhill, *Speaker Cables: Can you hear the difference?";* Stereo Review; Vol. 48, #8; pp. 46-51 (August 1983).

The Questionnaire

In past columns I have made sweeping generalizations about what people in this industry believe. Before I get myself into too much trouble, I'd like to ask you, the industry leaders, what today's conventional wisdom really is. To make this as painless as possible, we have set up our opinion survey as multiple choice but, of course, would encourage longer answers. You also will notice at the bottom of the survey we ask for some demographic information; running this data through a computer should result in some fancy percentages regarding who holds what opinion.

The following list of statements are from my first column (April issue) with the addition of a few others that were suggested by *R-e/p* readers. (See, we *do* listen.) If you don't see an answer you like, write in your own, or skip the question. The more returns we receive, the more the results will be representative of all of us. I apologize in advance if some of the questions seem leading, but we are trying to gauge opinion here; we're *not* trying to find out what everybody *knows*, only what everybody *thinks* (huh?).

You might like to photocopy these pages, tick the appropriate boxes, and then mail the completed questionnaire to me, c/o *R-e/p*.

Question 1: I prefer the sound of *digital* audio.
☐ YES ☐ NO

Question 2: I prefer the sound of *transformer* circuits.
☐ YES ☐ NO

Question 3: Gold-plated jacks, sockets, etc. *do* make a sonic difference.
☐ YES ☐ NO

Question 4: Speaker-wire size *does* make an audible difference.
☐ YES ☐ NO

Question 5: Power amp slew rate *does* make an audible difference.
☐ YES ☐ NO

Question 6: Negative feedback *improves* sound quality.
☐ YES ☐ NO

Question 7: Capacitors *can* cause audible signal degradation.
☐ YES ☐ NO

Question 8: (Multiple Choice) How many dB per octave crossovers sound better?

6 dB per octave ☐
12 dB per octave ☐
18 dB per octave ☐
24 dB per octave ☐
__________ dB per octave

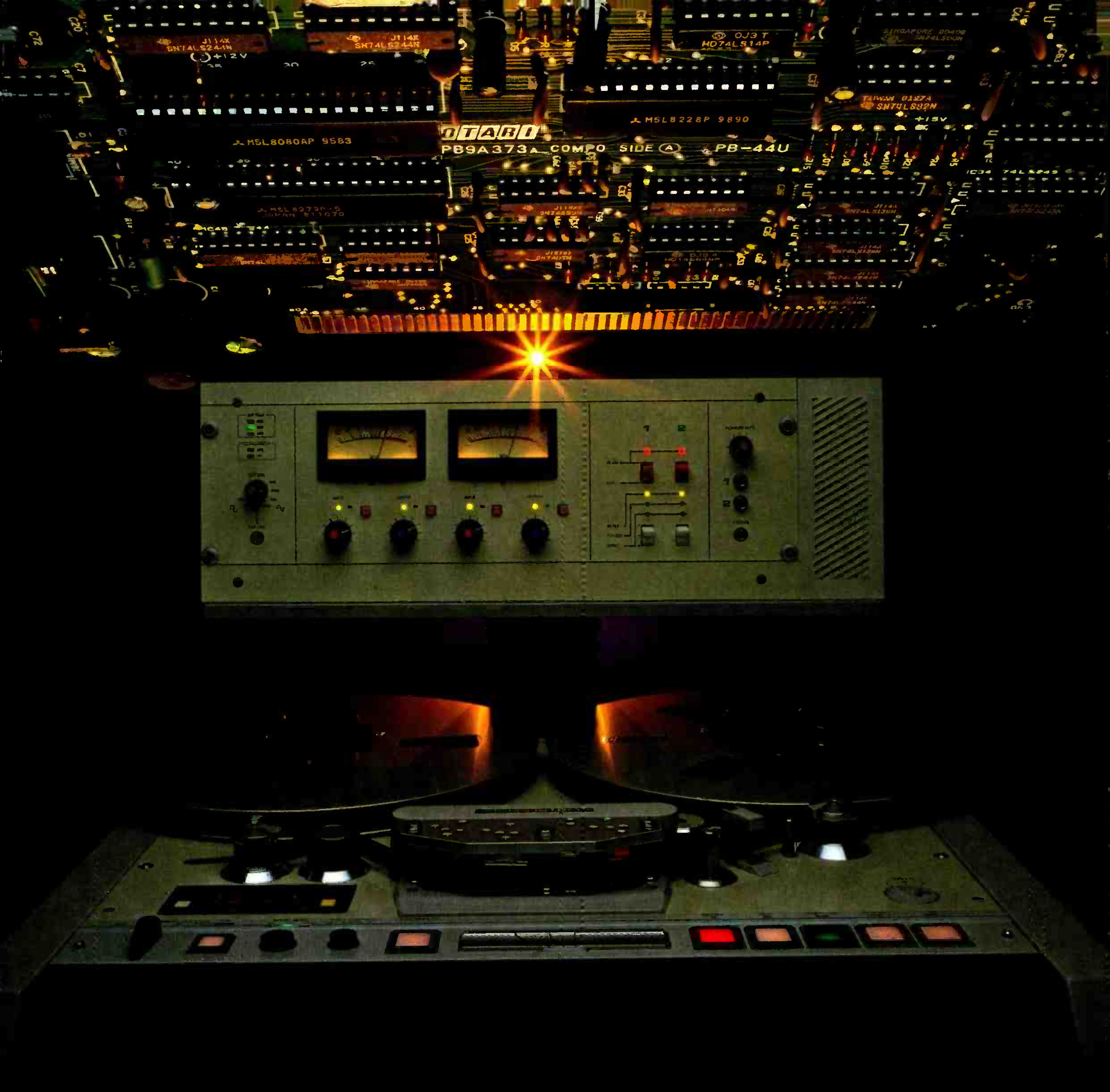
Question 9: Tube electronics *do* sound better than solid-state electronics.
☐ YES ☐ NO

Question 10: Discrete circuitry *does* sound better than integrated circuits.
☐ YES ☐ NO

Question 11: Moving-coil cartridges *do* sound better than moving magnets.
☐ YES ☐ NO

Question 12: Half-speed record mastering *does* sound better than conven-

— *continued on page 25 . . .*



AUDIO MYTHOLOGY

... continued from page 22 —

tional mastering.
☐ YES ☐ NO

Question 13: You *can* improve the sound of electronics by adding weights.
☐ YES ☐ NO

Question 14: The sound quality of low-level signals *can* be affected by the wire used.
☐ YES ☐ NO

Question 15: Unused speakers in the same room as driven ones *can* degrade sound.
☐ YES ☐ NO

Question 16: MOSFET Power amps *can* sound better than bipolar amps.
☐ YES ☐ NO

Question 17: +4 dBm systems *are* superior to -10 dBv systems.
☐ YES ☐ NO

Question 18: Dubbing tapes backwards *can* improve the copy's transient response.
☐ YES ☐ NO

Question 19: The type of jack or interconnect *can* make an audible difference.
☐ YES ☐ NO

Question 20: The amount of power supply regulation *can* affect the sound quality.
☐ YES ☐ NO

Question 21: Phase shift and/or absolute polarity *is* audible.
☐ YES ☐ NO

Question 22: Proximity of speaker wire to structural surfaces *can* be audible.
☐ YES ☐ NO

Question 23: Loudspeakers *can* image wider than their physical placement.
☐ YES ☐ NO

Question 24: Phase coherence (time alignment) of speakers *is* audible.
☐ YES ☐ NO

In the following question you may not be able to just check one box. For example, if you are an engineer part of the time, and a producer the rest of the time try and estimate the spread (i.e. 25% engineer/75% producer).

I consider myself a:
☐ **Golden Ear**
☐ **Meter Reader**
__________ **(other?)**

INDUSTRY INVENTIVENESS

Due to circumstances beyond our control, we are unable to publish James Riordan's column, "Industry Inventiveness in the Eighties," in this issue of *R-e/p*. Jim's column, however, will continue in the December issue. Our apologies to all — *Editor.*

ONE YEAR LATER: A Progress Report on the dbx Model 700 Digital Audio Processor

by Leslie B. Tyler, vice president of Engineering, dbx, Inc.

Readers will recall that dbx introduced the Model 700 Digital Audio Processor at the October 1982 AES Convention, in Anaheim, California. The 700 uses a proprietary analog-digital-analog-conversion technology that we call CPDM, Companded Predictive Delta Modulation. [For a detailed explanation, see the article written by 700 design engineer Robert W. Adams in last October's *R-e/p — Ed.*]

Since its introduction in prototype form, dbx has been doing extensive field and laboratory testing of the 700. This testing has resulted in several important modifications whose goal is the debugging and perfecting of the design. We now know that the product will easily meet the requirements of the most demanding program material. What follows is a summary of the changes:

Compander

Tests at various studios throughout the country revealed essentially that the 700's performance was as we had hoped: 110 dB of dynamic range, and a neutral noise floor. Most important, to quote many of the auditioners, it was "musical." We did discover, however, that very high-level, high-frequency material was recorded and displayed on the meters at a higher level than we liked to see. That is, for a given sound-pressure level, tambourines or wind chimes (for example) were recorded closer to the saturation level of the A/D converter than was equally loud lower-frequency material. The practical consequence for the user was that the meters had to be watched more closely than usual. We thought it prudent — a matter of good engineering — to make the design "bulletproof" under even the severest conditions if we could. Therefore, we changed the compander so that recording level was flatter as a function of frequency content, which renders the 700 even more immune to overloading than before, and makes the meters behave in a way users would be more accustomed to.

The earlier version, described in these pages a year ago, used classic amplitude-compressor/expander techniques to increase the dynamic range from around 70 dB (achieved through the use of our linear-prediction circuit from the typical 55-dB basic performance) to more than 110 dB. The new system, to improve the situation mentioned above, uses *spectral* as well as amplitude compression.

A spectral compressor is intended to reduce variations in spectral content, just as an amplitude compressor reduces the variations in amplitude. In the dbx 700, the spectral compressor shapes the spectrum presented to the A/D converter. It reduces high frequencies when the input spectrum contains predominantly high frequencies, and boosts them when the input contains predominantly low frequencies. The spectrum at the A/D converter input (the output of the spectral compressor) therefore is more nearly constant and, to the A/D converter, more "comfortable" to deal

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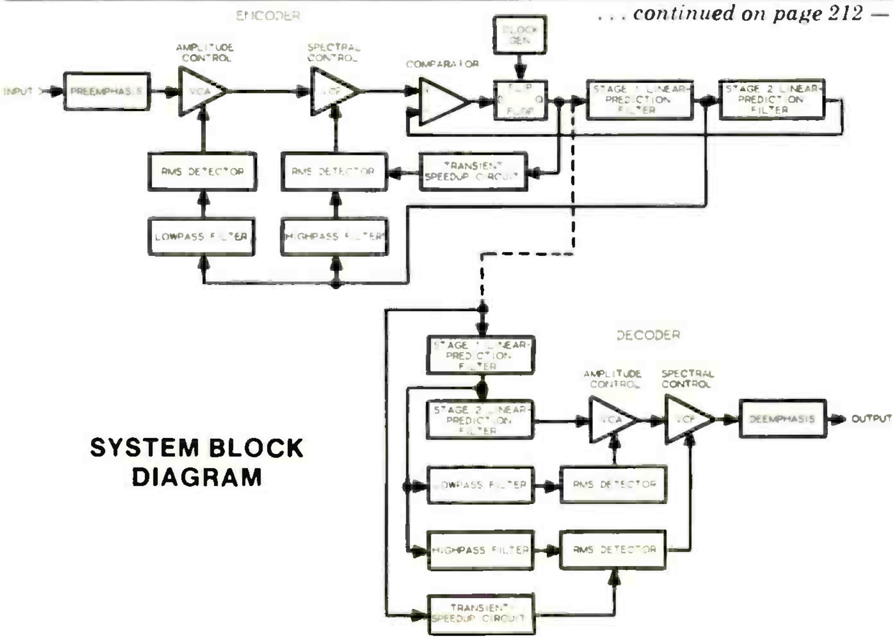


SYSTEM BLOCK DIAGRAM

All Photography by Kathy Cotter

VAL GARAY

Interviewed by Robert Carr

As a natural extension to his career as a musician during the early Sixties, Val Garay's love for music lead him to pursue the art and science of audio engineering. Starting in 1969, he apprenticed at the Sound Factory, Hollywood, under rock-recording legend Dave Hassinger (Rolling Stones, Grateful Dead, Jefferson Airplane, Seals and Crofts). After turning independent, Garay formed an alliance with another ex-musician, Britisher Peter Asher. The association produced monster hits for Asher's clients Linda Ronstadt (*Heart Like a Wheel, Prisoner in Disguise, Hasten Down the Wind, Simple Dreams, Living in the USA, Mad Love*) and James Taylor (*J.T., Flag, Dad Loves His Work*). Garay eventually became dissatisfied at the Sound Factory, and the inconsistencies attendant with moving from one studio to another. At which point he decided the best course of action was to open his own facility, Record One, located in Sherman Oaks, just north of Los Angeles, and which now serves as his recording home.

The following interview took place among the dozens of Gold and Platinum albums lining the walls in Garay's private office. After a few words on his recent accomplishments as producer/engineer with Kim Carnes (*Mistaken Identity*; 1981 "Record of the Year" Grammy Winner for "Bette Davis Eyes"), Randy Meisner (*One More Song*), Joan Armatrading (*The Key*), and the Motels (*All Four One*), a band that Garay also manages, the conversation turned to the opportunities and advantages to an engineer/producer owning one's own personal-use studio.

R-e/p (Robert Carr): It must be particularly convenient to have your own studio, which enables you to take the time to perfect each project you work on?

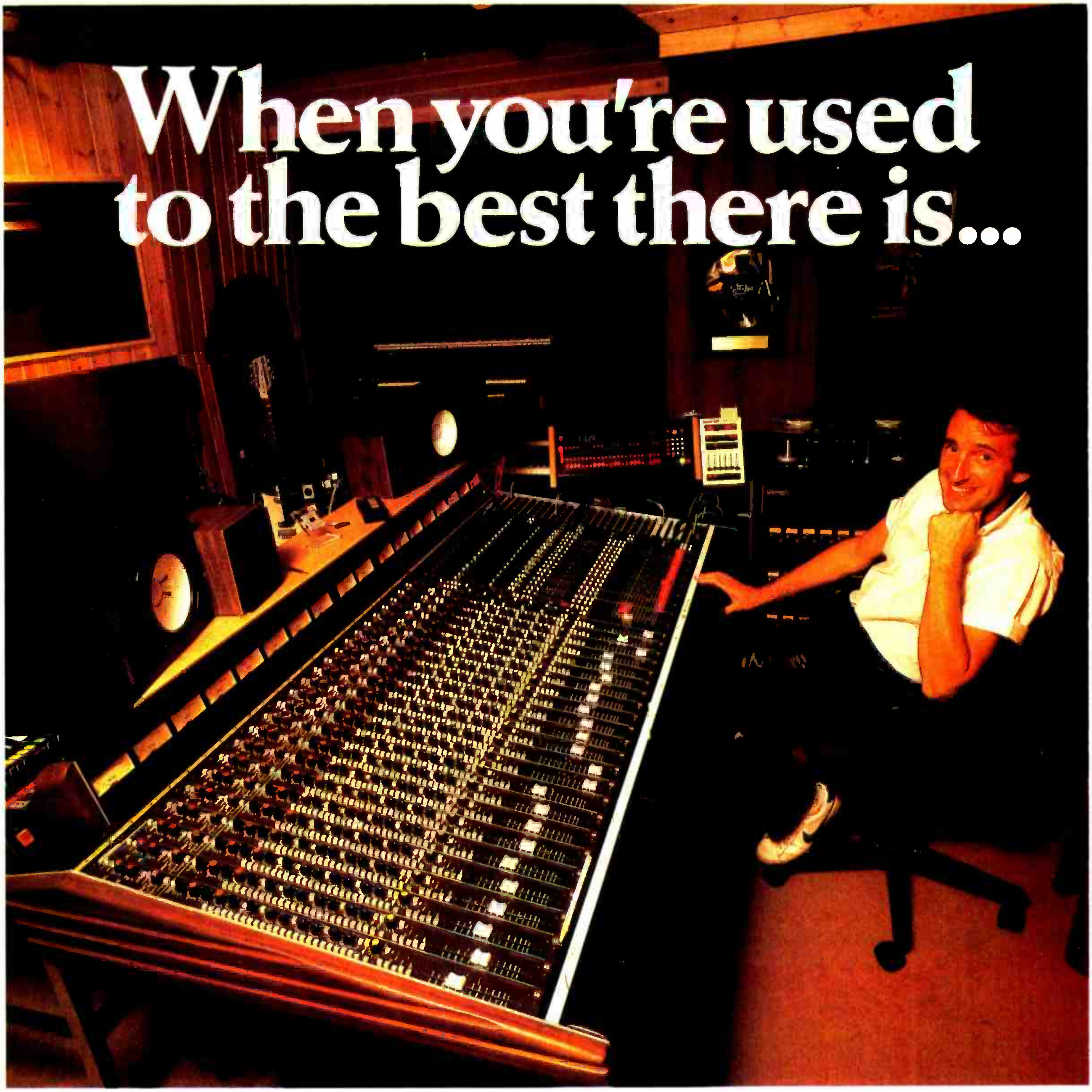
Val Garay: It is, and it isn't. Sometimes it's a pain in the ass, because you have to deal with the business end of owning a studio, which I'm *not* terribly fond of. I don't like to sit there with calculators and figure out the plus and minus side of the operation. I like to make records, which is a lot more creative, and pretty soon I'll start making a film. [A feature film based in part on Motels lead singer Martha Davis' life currently is in its development stages.]

Owning your own facility is kind of a necessary evil in the sense that if you subject yourself to a commercially rented studio, you subject yourself to someone else's taste — not only in terms of equipment and design, but also maintenance and other things. I was fortunate to spend the first eight or nine years of my engineering career in one recording studio [Sound Factory in Hollywood], and the rest of the time here [Record One]. I wasn't subjected to going from one studio to another. It's too unsettling for me.

R-e/p (Robert Carr): Is stability of that nature necessary for you to make a good product?

Val Garay: I think you perform better when you have familiar surroundings and equipment that you're used to working with. If you were a "body-and-fender" man, to put it on a mundane level, and you were wandering around the streets doing your work every day using different tools in different areas, I'm sure you wouldn't be as proficient as if you had your own body shop. It's basically the same thing here.

The only problem is that this is a two-



"You would choose a Soundcraft, too". That's what producer/songwriter Pete Bellotte told us.

And he certainly is used to the best. While he was co-writing and producing most of Donna Summer's hits he chose Soundcraft for his own studio.

"When we were recording Donna's 'Macarthur Park' and 'Bad Girls' we were amazed at the flexibility of the Soundcraft and in particular, the EQ. You get all the control you need out of a Soundcraft EQ without any harshness or distortion. The choice was simple.

"My major guidelines in selecting the equipment were space and quality. The Soundcraft 2400 and SCM tape machine met those criteria. With the split configuration of the 2400, it performs like a much bigger console. When I mix down I can use the monitor section of the console to return more tracks. I've got the capability of mixing down up to 52 tracks simultaneously and I don't have to stand up to get to the controls – they're all within easy reach.

"Once you're used to the best there is, you'll never want to compromise. And that's why I chose Soundcraft".

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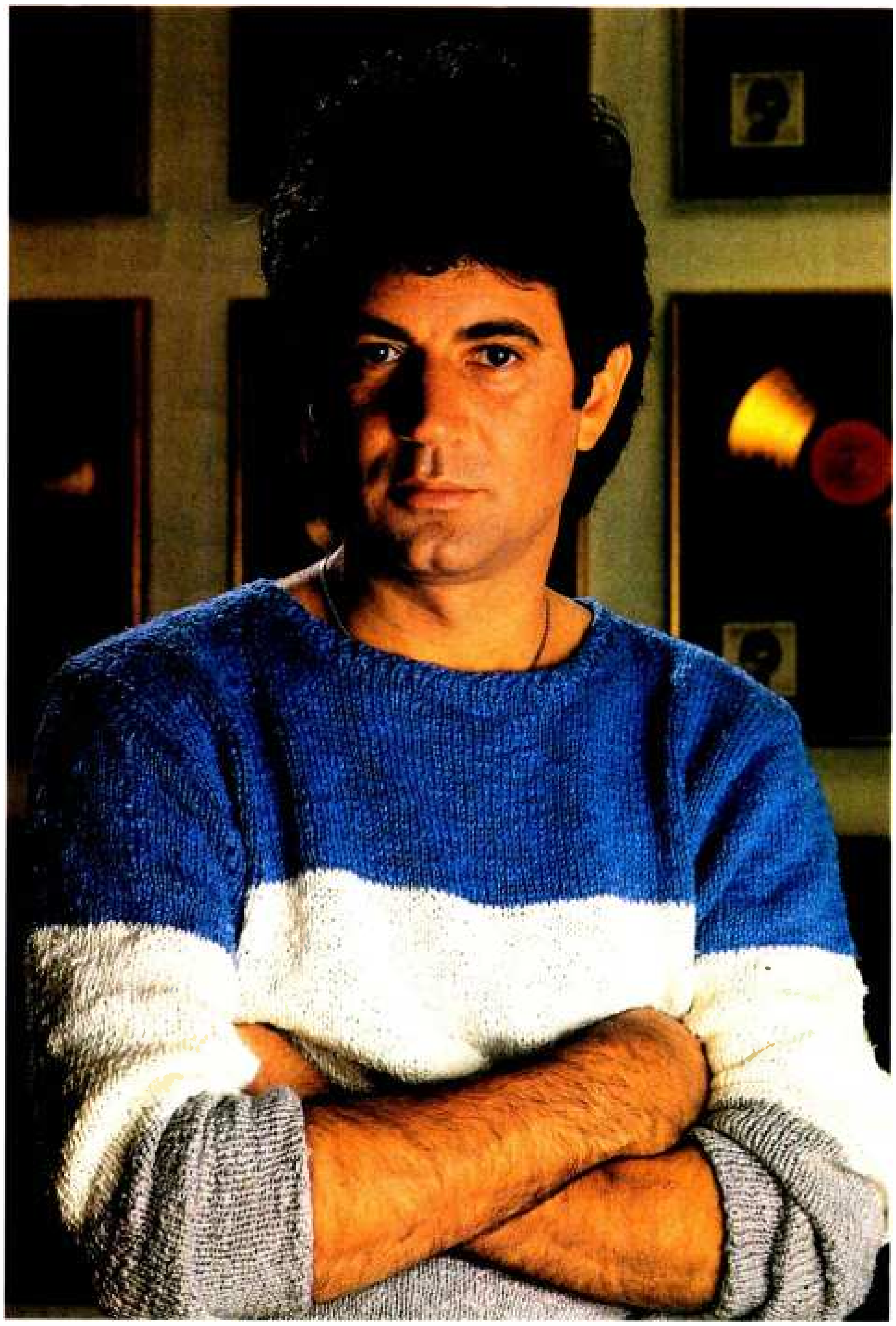
This Is
Val
Garay.

He Is The
1981
Grammy
Winner
For
"Record Of
The Year"
Bette Davis Eyes:
(Kim Carnes).
Musician.
Engineer.
Studio Owner
(Record One).
Producer.
Personal
Manager.



Val Garay Masters and Mixes Exclusively on Agfa Tape.

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The Mastering Choice of the Masters.

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For additional information circle #20

million-dollar operation, so it requires a *lot* of attention. And I'm not the only one who uses this studio. We rent the studio to a lot of clients, and I'm constantly having to book around other people. In all fairness, if I decided to work tomorrow, I couldn't bump Toto out of the studio. I'm basically a customer here, too; that can be frustrating at times.

R-e/p (Robert Carr): Couldn't you divest yourself of the day-to-day running of the studio, and put someone else in charge?
Val Garay: No, I can't. It's the same way that I make records. I have to concern myself with every fragment, or something starts to dissipate or disintegrate. If you're not in contact with what's going on, you can't catch it before it gets too bad.

R-e/p: I assume that kind of philosophy is what motivated you to become involved with both engineering and producing the projects you take on?
VG: I've been working this way for 15 years. I just wasn't successful as [only] a producer. But it's very difficult to try and hire somebody to engineer records when, in my mind, and I *don't* mean this egotistically, I'm one of the best engineers I know. How could I hire somebody else? All the really good up and coming engineers that I know are people I taught. And you know that you teach them everything *they* know, not everything *you* know.

Greg Ladanyi won a Grammy last year for the *Toto IV* record, and I taught him. I was just reading an article in *R-e/p* on Gabe Veltri [April 1983 issue — *Ed.*]. When I got perturbed at the Sound Factory at one time in my career and went over to Richard's [Perry] studio for about a year and worked, Gabe was my go-fer. Now I see him in his sweater and tinted glasses behind the console.

It would be very difficult for me to hire someone as my engineer, unless I worked with someone in my peer group. I could work with [Bill] Schnee, because we came out of the same school in the same time frame. But when you have somebody else to deal with, you have another personality, another X-Factor in the formula. That tends to dilute the process sometimes. Whereas right now, I don't have a whole lot of conversation with my engineer about how I want to do something, because he *knows* how I want to do it, since he is *me*.

R-e/p: A lot of producers don't like to handle both functions for the same project, because they feel they'll be missing some production aspect while they're working with the equipment, or vice versa.
VG: It can be hard. But here's how I do it, which is actually pretty easy, because I've figured out a method that works. I spend an immense amount of time rehearsing, which is why I built a rehearsal studio in here [Record One]. That's when I sort out the musical part of the record-making process — the instrumentation; the arrangements; the basic architecture of the song [see accompanying sidebar].

The ratio of rehearsal to recording time is about two-to-one. If we spent eight months making a record, two-thirds of that was rehearsing, and the other third recording. We figure everything out in absolute detail and make cassettes at each juncture as we go along. I could play you cassettes of the Motels' album [*All Four One*] that shows one song passing through four stages of arrangement. Sometimes we'll get into the studio, cut the tracks, not get it, come back to rehearsal, and work on the arrangement even more.

"You have to motivate yourself. That's how I do it — with fear."

By the time we get to the studio, I'm *thoroughly* familiar with the song. There are so few changes made while we're recording that I can become an engineer and get a sound that I like. Once I've accomplished that, there's really nothing more to laying it down than cutting a vocal, and I can do that without even thinking about it; my hands respond unconsciously to how my ear wants to hear the vocal track. I don't even look at VU meters anymore. I'm totally conscious of the music when it's going down, and I can tell a great take from a bad one *instantly*.

I also make notes. I keep a loose-leaf notebook for every group I work with. Here's the Motels'; this book represents the last album we did. [Garay holds up a black binder, and opens to a page about half way into the book.] If you look at "Only the Lonely," for example: this is the lyric sheet [flips page]; I have the date on the top of each sheet. These are the fixes we did on the vocal; the numbers of the takes with little one- and two-word descriptions after each one. As the track is going down I make notes: "CT" equals complete take; "FS" equals false start, etc. [Sample comments: "bad sax"; "good take"; "the run-through was good in spots"; " still some mistakes"; "end is not tight"; "magnificent from solo on"; "the last hit was perfect."] Here is my star system, actually stolen from Peter Asher: two or more stars means that the take was *really* good.

I keep pretty accurate notes of everything that I've done on every record. Sometimes the notes get more excessive or less depending on how hard it is to cut.

Here's Kim's album, *Mistaken Identity*. [Garay pulls out another binder from the pile, and opens to a page.] "Bette Davis Eyes" — that was the first complete take. Then in the back is usually the songs that didn't make it. "The Lover" didn't make it, obviously. Neither did "New Orleans Ladies," "Here Comes the Bad One," "Good Friend," "Games," "If You Don't Want my Love"; these are songs that never made it as we were working on the album.

R-e/p: Did you spend time pre-producing all these songs that didn't make it?
VG: We rehearsed them. The ratio I've found in the past is usually three or four to one, meaning 30 to 40 songs to get 10 finished ones. For every three or four songs, you'll get one that not only suits the artist, but is also strong enough to use on the album.

R-e/p: Do you keep those rejected songs for use in the future?
VG: It's a nice idea, but unfortunately it never works. If they are not good enough for this album, usually they won't be strong enough for any one.

R-e/p: You're really playing the numbers. You start with a lot of songs, and slowly weed them out until the good ones turn up?
VG: Not necessarily. When I started the new Motels' album in January 1983, we had three songs. Three became one; one became none. Then we started over again. We have all 10 songs now [July 1983].

R-e/p: If you do spend six to eight months or a year on an album, is it cost-effective to do everything yourself, assuming that your time is worth quite a bit of money?
VG: Hiring someone else to do those things would not change the time frame at all. We'd have to rehearse just as long, and it wouldn't change the engineering.

I originally did all the pre-production out of fear of not being prepared in the studio, and not being able to make things sound good. But, in reality, that kind of time is required to do a good job, regardless of whether you're engineering or not. One sort of facilitates the other anyway.

I make most of my records live with very few overdubs. I think that records are better that way, especially if you're

Production Viewpoint
VAL GARAY

working with great singers, which I have had the good fortune to do.

Yet certain singers thrive on the overdubbing process. I've never seen a great singer, who overdubs his vocal, sing a vocal part from top to bottom, and use 98% of it. The minute they get into the overdub design of doing vocals, they'll do 8 takes and comp [compile or combine] 8 to one track, then do 8 more and comp them. Basically what they do is use their ear as a singer to pick what they sing best, and sort of assemble the finished vocal track mechanically. In the end, it usually sounds like they sang it from top to bottom.

Don Henley does that very well — although I don't know *why* he does it because he's a great singer. In fact, all the Eagles did it that way for years. Jackson Browne does it the same way. They get as far as comping syllables. "Well, the *t-h-e* of that word is a little flat." So they'll switch at that point to another vocal track that has that syllable a little more in tune. The layman can't really hear all these comps. I did that with Randy Meisner's album; there were a million switches in that.

With the Eagles, [Meisner] was used to singing in only one register, which was really high. But for a solo record, where you're the lead singer, you have to cover all the areas. His lower ranges were a little more tentative, and he would sing out of tune more often. In order to get it in tune, we had to do the vocal tracks that way.

But when you have a singer like Martha [Davis, of the Motels], Kim Carnes, Linda Ronstadt, or James Taylor, those people are *great* singers. They have great intonation. The best vocal performances I ever recorded with Linda were the live ones with a few fixes — you fix one word here, and one word there. "Blue Bayou" was live; "Ooh, Baby Baby" was live. In fact, that whole record was. Also, "Bette Davis Eyes" by Kim was totally live.

R-e/p: I remember reading a couple of reviews about Linda Ronstadt's album to the effect that, because the recording sounded so perfect, the critics thought it had been "produced to death." How do you react to such comments.?

VG: The pre-production was really good. The interesting thing is that Linda never learned the songs until she got in the studio. She would sort of sluff her way through the rehearsals. The band would learn the songs, but she wouldn't even know the lyrics most of the time — she'd be reading from a sheet! But she's such a great singer that she can evoke emotions that sound like she's torn. She'd usually learn the lyrics in a couple of run-downs in the studio.

Martha [Davis] is a great singer. When you have someone that sings as well as she does, and a band that's got

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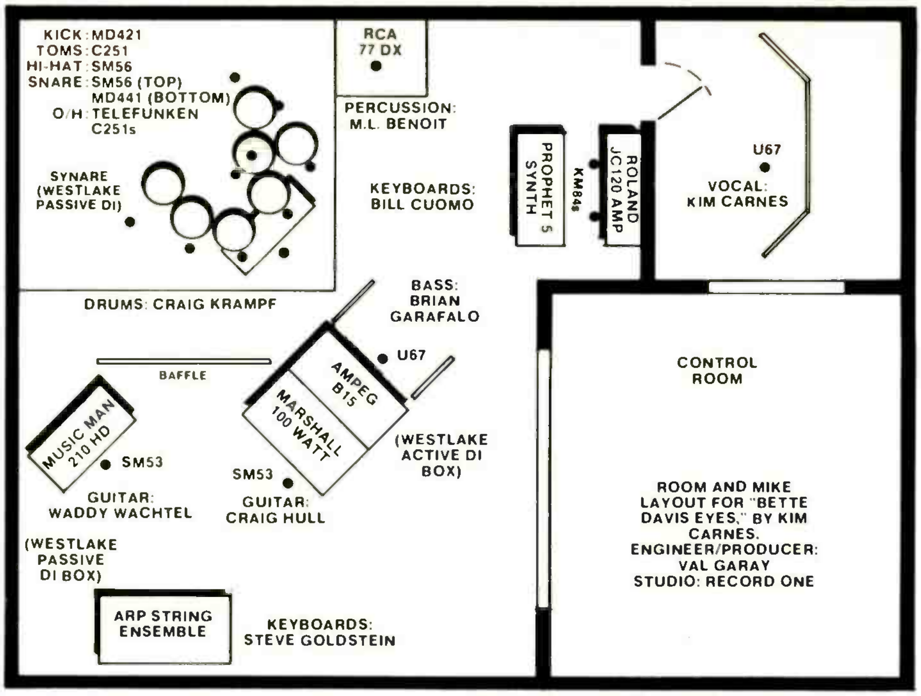

ROOM AND MIKE LAYOUT FOR "BETTE DAVIS EYES," BY KIM CARNES.
ENGINEER/PRODUCER: VAL GARAY
STUDIO: RECORD ONE

The Transition from Pre-Production to Studio Sessions

Pre-production starts with the set-up," Garay considers. "I let the band choose whatever makes them comfortable. Their rehearsal arrangement doesn't necessarily have to duplicate their normal stage or recording studio locations. As a rule of thumb, the drummer usually sets up in the center of the room, because everybody is listening to him. Then the bass player puts his amp near the drummer, so they can play easily together. From there it's pretty much up to the band members. A semi-circle seems to work, but the rehearsal room is small, so we could put anybody anywhere.

"As far as levels go, the band pretty much figures that out for themselves too, because there's an instant relationship among the members of a professional band. To hear each other in the room, there *has* to be a balance. If the guitar player is six times too loud, then all you hear is guitar and the other players tell him to turn down. But once you get the balance, you can stick one mike in there, open it up, and you're ready to make a work tape.

"I like to make recordings of each arrangement as we go along. I use just one little cassette machine, with one little microphone. I could play you work cassettes of almost every song on every record I've made so far. You'd be amazed at how much you can hear on those tapes. It's very close to the actual recording in the studio.

"The cassette tells you whether an arrangement works or not, because you can listen to it over and over. It tells you whether parts, rhythms, and everything else are the way they should be. The great test is how the song wears, and for that you *have* to keep listening to it over and over. The old adage is: "If it has legs, it will walk." What they mean by that is if everything about the song is comfortable, it will keep going. If not, it starts to grate. And it's either the arrangement or the song that grates on you. Once that happens, you have queries. And once you have queries, you start delving back into the song to find out why. I say that either the arrangement goes away immediately, or the song goes away in a period of time.

"When we go into the studio, I like to cut live — everything at once. I mike everything close for isolation, and also put very loud instruments, like distortion guitar parts, in separate rooms. [Record One features three acoustically treated recording areas — a main studio, and two smaller adjacent rooms — as well as the control booth, and various live rooms throughout the complex that are pressed into service when needed.] To make the separate tracks blend back together, I run feeds to two PA speakers in the rehearsal studio. I have two Neumann U-67s that I can move anywhere in the room, or right next to the cabinets, for any desired effect. I just open the microphones up, and add them to the original sound at the board.

"I don't really use a lot of effects other than the natural room ambience, when I want to change something. I like to get nice, big, warm, fat, *punchy* sounds. If you want an effect, you can warp anything with outboard gear, but you can't make anything sound big, fat, warm and punchy if it doesn't *start* that way.

"I guess you could say I'm a purist, but don't confuse that with traditionalism; a traditionalist I'm not. If there's a sound out there in the studio, *that*'s the sound I want to get on tape. I would prefer to play with the guitar player's amp and get the sound at his station, rather than attempt to manufacture what's needed in the control room. All I try to do is capture what he's got. In essence, the secret is that the studio and all the equipment must remain transparent to the overall process of recording." ■■■

If you examine the recording consoles available today, you'll notice many models that probably meet your needs in one form or another. In fact, if you're considering the purchase of a new audio console in the near future, you'll no doubt have a difficult time making a choice.

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So when you're evaluating several available designs, you can't really make a choice based on specific design aspects or specifications, since all modern consoles claim to be ruler flat, quiet enough for digital, distortion free and, of course, come complete with 24 buses. BIG DEAL.

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In its standard configuration, the Series 34 comes equipped with 28 input/output modules in a 32 input mainframe of exceptionally rugged construction and features four band equalization, high-resolution metering and balanced transformerless outputs whose clip level exceeds +27 dBu. The console is available with or without its computer-controlled VCA package featuring Sound Workshop's ARMS console computer.

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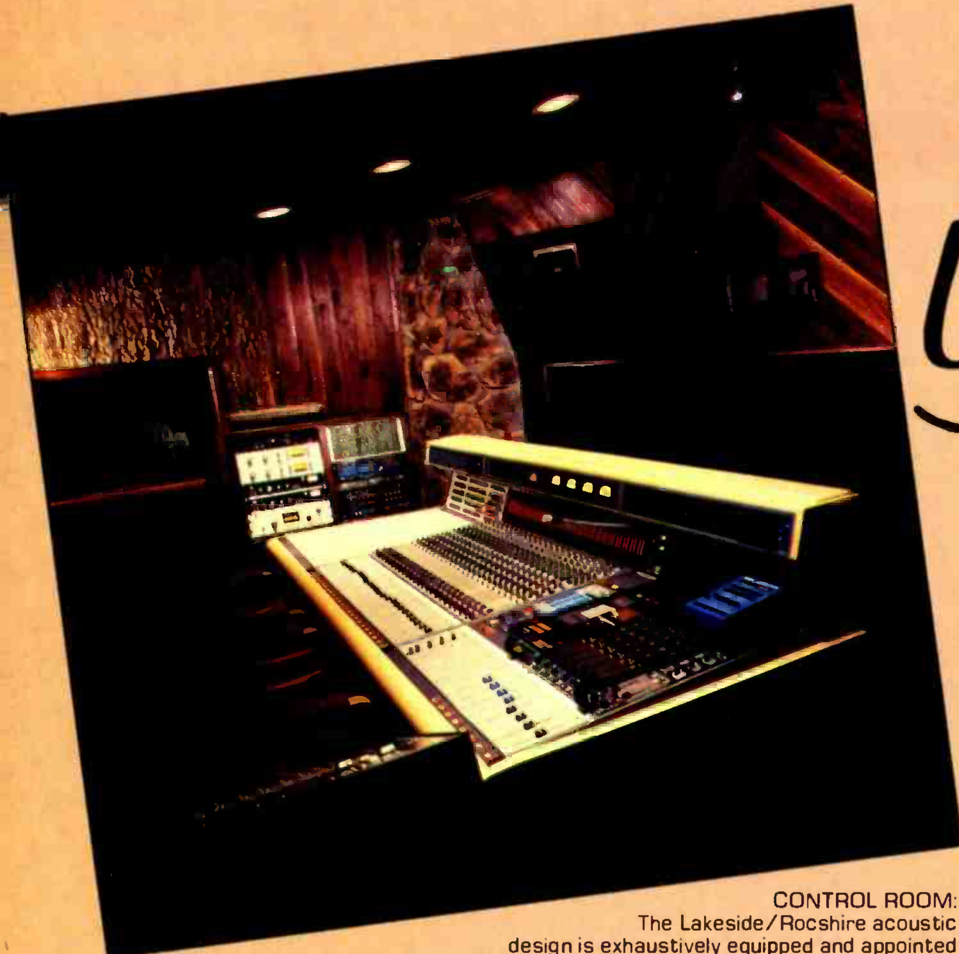

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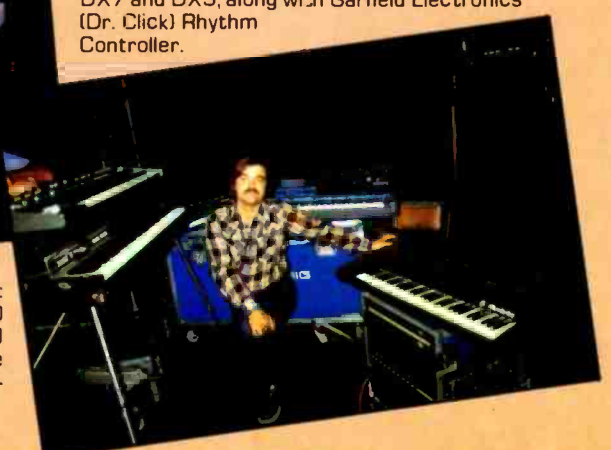

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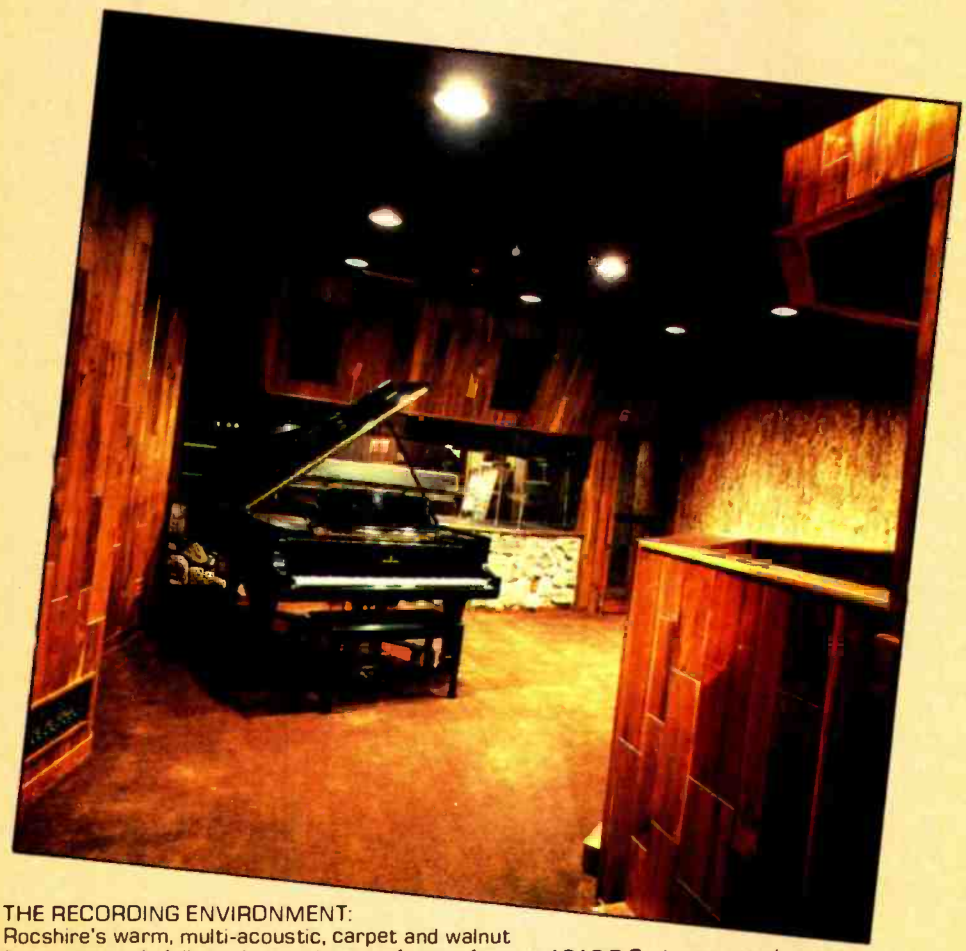

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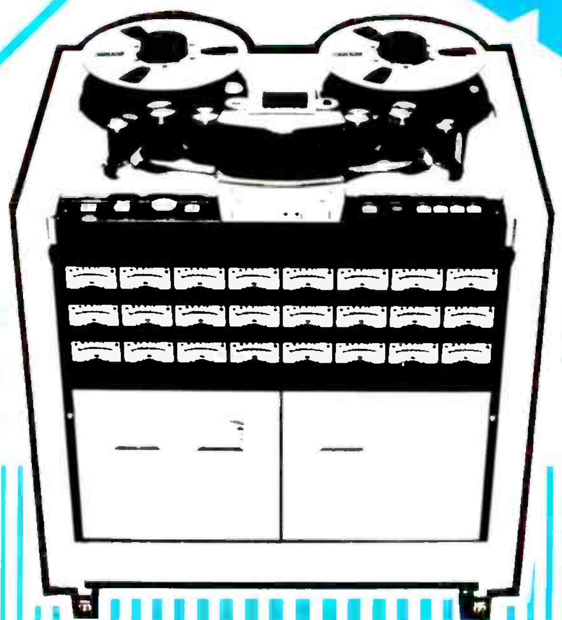

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For additional information circle #23

Production Viewpoint
VAL GARAY

— continued from page 32 . . .

the tune down — and they're interacting in a live-performance situation, even on record — it's much more real and emotional, and more moving, when it's all going down at once, and one person is playing off the other. If you have a strong drummer that doesn't move if the singer rushes or drags, then the track stays *steady*; the singer is singing and the band is following the singer, instead of a singer following a music track that's [already] laid. It's a whole different method. That's why Elvis Presley records made in the Fifties still hold up; they were done live.

R-e/p: You work with these artists for such a long time during pre-production and recording. It must be inevitable that you develop a close friendship with them after a while. In a way, doesn't it become harder to be critical of their work?

VG: It becomes easier, the more familiar you get with them, because the barriers and defenses go down. It's easier for me to be frank with Martha three years later, than it was the first month, because: *A*, I was afraid of hurting her feelings; and, *B*, afraid of what she was going to think of me. Is she going to think I'm a tyrant? No, the more familiar you become, the more open the lines of communication. You're more comfortable with the person, and there is less and less need for dialog. She knows what *I* want from her as a performer; I know, hopefully, what *she* wants, and we get to the point a lot quicker.

R-e/p: I would also think that it provides you with an insight into knowing when to kick them forward, and when to dangle the carrot in front to get them going.

VG: Absolutely. I've known her so long that I know when to say it's over; go home. Sometimes it's five o'clock at night; sometimes it's three o'clock in the morning. I know when the productivity level has peaked. That's when I go, "Good night. See you tomorrow."

R-e/p: I noticed that you tend to rely on the same session players for most of your dates. Does that stem from the same sort of philosophy . . . that you know them so well there's an extra efficiency?

VG: Sort of, but I think it has to do with more than that — a love affair with a great player. I'm sure that just as directors fall in love with actors and actresses, producers fall in love with musicians. I don't mean in a sexual connotation, but on an emotional level. When I first heard Russ Kunkel play drums, I was in *awe*. And he was a young man just starting out. But he had that *thing* that when you hear a great drummer, whether it's in the early raw form, or the finished polished form, you just *know* when you hear it. At least I do. So I worked with basically the same 10 musicians for 10 years.

When it came time for me to make a break with [producer] Peter [Asher], and start producing on my own, I knew it was imperative that I build my own little group of musicians, as opposed to using his. His were used to his method of operation. Although I learned a lot from the man, I wasn't going to do it the same way. That's when I started looking for the guys I wanted to use.

It's hard, too, because when you've dealt with the Waddy Wachtels, and the Leland Sklars and Russ Kunkels of rock and roll, you've set a standard that is pretty hard to duplicate. But I did, although I still go back and use Waddy from time to time.

R-e/p: What do you look for when selecting musicians for a session?

VG: I guess my own taste in musicianship. I know very few musicians who are feverish readers — playing noted parts that are written out. They can read their way through a rough chart, because most of the stuff we write out is just chord charts to give the people a guide to follow. I look basically for the *feel* they have for playing.

R-e/p: Many producers and engineers prefer not to work with the same people most of the time, because they feel that they reach a certain point in their careers where it's difficult to remain creative.

VG: That happened with the old group of musicians I worked with when I was with Peter Asher all those years. But [deciding] *when* it happens is not that clear cut. It's not that suddenly they don't become creative anymore, because their wonderful talent doesn't go away. It's just that you fall into a rut. It's like Steve Garvey playing for the [LA] Dodgers all those years, and last year he wasn't playing that well. Then he goes to San Diego, and he's killing them.

The same thing happens with musicians — familiarity breeds contempt. The temptation is to start getting lackadaisical. I know I can get a good drum sound on Russ Kunkel without turning up the speakers. I could leave them shut off, EQ them, balance them on VU meters, and know it would sound great, because I've worked with him that long. When you get to that point, you lose the *fear*.

When I make records, I operate under a fear premise that this project won't sound good enough, won't feel good enough, won't *something* good enough. It's fear. If I sit there and kick back, knowing I can get a great sound on these guys, because they're all going to play great, I've lost that hungry, street-level edge that got me here. That's what becomes difficult in terms of creativity.

Here's the difference; you've got the Phoenix Sun Devils and the New York Yankees. I'm sure there are days when the New York Yankees do not feel like playing baseball, but they do, because they're professionals. The same thing holds true in this business. When you're a professional, and you're good, you're respected, and you've reached a certain level of proficiency, you then have to figure out how to motivate yourself day after day. I have trouble with it. I've been sitting in a control room for 15 years looking at a pair of speakers. It's hard for me sometimes to go in there when I would rather be out in the sun sailing to Catalina, or playing gold at Riviera. I have other interests. But I have to get that fear of, "Is this going to be a hit record?"

Well, it's not going to be a hit record if I don't work on it. And it's not going to be a hit record if I don't put into it what I put into the last one.

You have to motivate yourself. That's how I do it — with fear. There's that guy right behind me; he's right on my heels. Until I decide to move into another area, I have to keep motivated. I have to keep up with the technology; keep my ears and eyes open all the time.

R-e/p: Other than the fear, are there other little games that you play to persuade yourself to look at the project a little bit differently, and to uncover new avenues?

VG: Yes. There's pressure . . .

R-e/p: Under pressure, wouldn't you fall

Production Viewpoint
VAL GARAY

back on the proven techniques and tricks you know work to get the job done?

VG: No. Well, there is a certain formula that is ingrained in all of that — what I call the basic foundation — that I live with. I never get rid of that. When my foundation was assembled in terms of making records, it was concrete — it was *solid*. I know what works. I also know the key to any record is a song. So if I do my homework in the song department, I can produce it in terms of the "production." Maybe not as well as the last record; I can sluff off in terms of the arrangement. But if it's "Every Breath You Take," I don't [care] if you cut it on a cassette machine in a phone booth in Tahiti; it's a *hit*. So most of the work I do is basically in the song line-up.

For the rest of the job, I'm fortunate. I learned from a great teacher how to make records; I know how to make them sound great. I can do R&B; I can do rock and roll; I can do country music . . . pop music. I've done all of them successfully. I've had a well-rounded career doing acts like that, so it's just a matter of finding things that I'm comfortable with.

R-e/p: Up until now, we've been discussing primarily rock projects. Do you feel you've become something of a rock specialist?

VG: I think that was done out of self-defense. By the time I was done with eight or nine years of Linda and James, I was stamped as the engineer for country-pop — the "California, surfboards, and tuna-fish" engineer. Oddly enough, my roots were always in rock and roll long before I ever did anything with Linda and James. So, out of self defense, I went after projects with more of a raw, rock-and-roll edge to them, to prove to people that I could do that type of music. That's sort of where I've been for a while.

R-e/p: Do you really feel that you've gotten stuck there?

VG: No. Not at all. I like it. I like to take acts that are slightly off-center — not main-stream pop acts, but slightly off-center, rock-and-roll acts — and make them mass-appealable. All the acts I've worked with since I started as a producer had not turned the corner and become big, successful recording acts before I worked with them. They were all a little bit *off* in terms of their style, or their singing, or their sound, or whatever. I figured out a way to make them acceptable to the masses.

Kim Carnes had made six albums before I started working with her. She had gone from the beautiful, southern California singer/songwriter, to the woman with the raspy "Rod Stewart" voice doing a song called "Bette Davis Eyes," which is about as off the wall as anything you can ever write.

Martha [Davis] had made two albums before I worked with her — neither were successful. Everybody knew she had the potential. She was sort of the "Los Angeles, New-Wave hope." People had assigned her the slot of *heir apparent* to the throne of the female, LA, rock-and-roll star. It hadn't happened. Again, I think I helped figure out a way to make it work.

R-e/p: Of all the albums I listened to, Kim Carnes' Mistaken Identity *sounded the most commercial. It had a Top-40 sound to the album, whereas the others — the Motels, Joan Armatrading, and Randy Meisner — didn't.*

VG: Joan's record is pretty *avant garde*. I only did two tracks on that [album],

"I was stamped as the engineer for country pop — the 'California, surfboards, and tuna-fish' engineer."

and those two were probably the most commercial. The Motels' album sounds really commercial to me, and considering how well it did sales-wise . . . "Only the Lonely" is, to me, the classic cheek-to-cheek tune. I don't really know what you mean. "Mission of Mercy" was a great AOR rock tune.

R-e/p: I can describe it more in terms of colors. Mistaken Identity *had a very light color to it, in the sense that you might hear it on a middle-of-the-road station. The other material comprised darker shades of colors.*

VG: Right. Martha is a very dark writer. Kim has a lighter side to her that is really pleasing. To me, her real strength as a singer lies in the fact that she has this wonderful sensitivity. A song like "Mistaken Identity," or "Bette Davis Eyes," is amazingly captivating, because she can evoke both of those emotions out of you. Whereas, you listen to a song like "Break the Rules Tonight," which is her screaming her ass off, and the guitars going "gggrrrk, ggrrk, ggrrk," that's good, but it's not as believable to me as the other side of her.

R-e/p: So part of your job is to establish a direction and identity for the artist, and have them remain credible within that identity?

VG: Absolutely. The toughest part of the job is to have them not lose credibility in their minds and, at the same time, be accessible to the masses. You don't want them to feel like you are selling them out. You have to show them you're on *their* side and, at the same time, strike a happy medium between the absolute *avant garde* side and the mainstream, pop medium, which sometimes tends to be a little bland.

Critics talk about an artist selling out when they [the artists] get successful. The reason that all the *avant garde*, hard-core people think you are *selling out*, is because you appeal to the masses.

R-e/p: You're no longer something that they discovered?

VG: Right. I watched Hoyt Axton completely berate and belittle Linda for *selling out* when she made "Heart Like a Wheel," because he was this hard-core country singer. She worked her ass off thinking that she was making a sound, artistic endeavor. Because it sold 2½ million records does *not* mean she sold out. But, to him it did, because she was no longer *his* discovery.

R-e/p: Is there a process that you go through to define an artist's personality, or is that a difficult concept to put into words?

VG: It's not that nebulous; it's pretty real. The quickest and most efficient way of doing it is through songs. If the artist is a writer, they write great songs, and not as great songs. But they are not always the best judge of which ones are the great ones! *My* job is to find the great ones.

It's a funny kind of "push-and-pull" process where I'll listen to five tunes and say, "This is a good song; these four aren't." They'll go, "Well, I really love this one, too." And I tell them it's not really that good, but we'll work on it. Then we work on it, and it's still not good, and I say, "Forget it. Let's off it." And they come back with, "No I really love it. We *have* to keep working on it!" So we keep working on it. Sometimes you keep going over and over and over and finally you have to say, "Forget it! It stinks! Next tune!" Or you may get it. Suddenly it all comes together.

We had a tune like that on this album with Martha. We started cutting in February and finally got it in . . . [flips through notebook of Motels' sessions]. That's a good note, huh? [Garay points to a qualitative note about a take on one of the pages.]

R-e/p: Horrible! [*Laughter*] *Do you show the artists this book as you go along?*

VG: It sits in the control room next to me. They always come in and look to see what I said. That's the first thing they do to find out if they got a take or not. [Continues to flip through pages.]

Here you go. We started it 4/14/83.

Production Viewpoint
VAL GARAY

This is the first time we cut the song, so we know they've been rehearsing it for a week or two weeks in front of this. And we cut it on the 15th [flips through pages]. That version sat around for a while, and then we realized that it wasn't right. Then, the 24th of May: a new version. That didn't fly. We changed the arrangement and we cut it again on 6/6/83. Then 6/7/83; that's when we got it. The eighth take. Figure that's almost two months on that one song to get a recorded version we liked.

R-e/p: Did the song change that much during those eight weeks?
VG: Drastically. Four completely different versions. I still have them on cassettes.

R-e/p: Would you say that a lot of artists really don't know who they are? Or don't have a really clear picture of themselves?
VG: Always.

R-e/p: So the whole idea during production is to cut through the illusion of who the artist thinks they are, and find the real *self?*
VG: I don't tell them anything. I just help them find what they feel comfortable with, and what I think is an acceptable mode to the general masses, as opposed to a select few. The Motels were successful previously on an underground basis, because they made albums that were an *avant garde* kind of collector's item. That's about all they were. They had some great songs on there, but they just didn't come out.

R-e/p: There's only a finite number of things you can do with a song . . .
VG: . . . it's *endless*. To give you an example: you have 10 songs on an album. When you go to sequence that album, what are the multiple number of ways that you can arrange those songs? Millions; about 3,856,000 and change. Yet if you gave me 10 songs and asked me to sequence them, after I became familiar enough with them I would say that every time I would get that sequence into four or five logical combinations.

R-e/p: The songs tell you where they want to go?
VG: Kind of, but you also have an endless supply of options. And you never get to a point where you have exact figures. The technology changes so fast; the styles of music change so fast. Basically we're looking at the parameters of tone and time. Time based on . . . this year it's more of a mechanical sound with mechanical drummers; synthesizers synchronized to the mechanical drummers with sequencers. Five years ago it was something else.

And song form has changed, too. It's no longer verse-chorus-verse-chorus. That's changed drastically, based on the boredom of familiarity. I can see Martha's writing style change from the way John Phillips would write a song for the Mamas and Papas, which is classic Gershwin or Hammerstein kind of verse-chorus-verse-chorus-bridge-etc.

R-e/p: What about putting the arrangement together? Let's say on a chorus, where the harmony comes in for the first line, solo voice for the second line, harmony again on the third line . . .
VG: Well, that's pretty much not going to change a lot, because that's basic song architecture. You want the beginning to be intriguing, and draw you in, but you want it to get bigger as it goes down the road.

Now there are a lot of "no-music solos," which five years ago you didn't hear. I think I produced the first big hit with the first no-music solo in "Bette Davis Eyes." That had no noted music in the solo; it was just a riff rolling over and over. That's happening a lot. The Police record ["Every Breath You Take"] has almost no solo in it. A couple of tunes on Martha's new album have no solos.

R-e/p: Obviously there's a constant striving to throw at least one new thing in every song?
VG: I think that happens by itself. Every time I've ever said, "Okay, this time we're going to come up with the great new sound," it turns out to be junk. The Synare [drum machine] that I used on "Bette Davis Eyes" and everybody copies now, is a good example. It was a complete accident on my part. Nobody went, "Let's come up with this great sound that everybody will copy." [Drummer] Craig Krampf went out and bought one of these little things that you hit and it goes [Garay does a Synare impression with his mouth]. He's sitting there trying to put it in every song we rehearsed for two weeks. Finally I told him, "Will you throw that thing away? It sounds like a garbage can lid."

Then we started working on "Bette Davis Eyes," and he played it in that. I went, "Wait a minute! That works!" You've *got* to be ready to try things. I worked on the *Summer Breeze* album with Seals and Crofts and [engineer] David [Hassinger] at the Sound Factory. About eight months later they did a live show where the actual live take of "Summer Breeze" wasn't that good. Louis Shelton said, "Why don't we cut to the downbeat of the studio 24-track, and to the last beat of the live tape? That way we'll be using the original 24-track for the song." I said, "No way. It'll never work in a million years. Forget it. You can't do that." "Try it," he said. "No!" "Try it!" "Okay!" Cut. *Perfect*. Worked great.

Never say *never* — you have to try everything. I can say *no* after I've tried it. But all the time I'm in the studio I'll say, "Why don't you try this thing?" and I'll get resistance.

Working with an artist is the hardest thing to do. It's like raising children. I don't have any, but I've been around enough of them in my lifetime to know that when you're a parent, the hardest thing to do is not to impart your values and your personal judgement on the child, who is very impressionable and wants to learn. You want them to be themselves a little bit. You don't want to keep saying, "No. You can't put your pants on that way. No, don't sit that way on the couch." Pretty soon, they become a puppet to you, and your feelings and values. But when you let them be themselves, they are amazingly honest. That's because they aren't inhibited; they have their own method of thinking and operating.

It's the same thing with an artist. They are lovable little children in a lot of ways — that's what makes them so vulnerable. So the hardest thing is to try and help them out of the womb, but not smother them. You've got to let them grow on their own, and it's hard; it's painful a lot of times. I put a lot of work

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Production Viewpoint
VAL GARAY

into an artist and a project and a career and, as they grow and become less and less dependent on me, it hurts. I've nurtured them, held their hand, put the band-aids on their knees . . . all the things you go through as a parent. When they become independent, it becomes difficult. But at the same time, there is the satisfaction of being the proud parent standing there at graduation when they're accepting their *cum laude* award. It's mixed emotion.

R-e/p: If one of your artists came to you, expressing the wish to work with another producer, how would you react?
VG: Actually, that sort of happened with Kim. I made a really good record with her that sold [in] unbelieveable amounts, and she decided after we made the second record — which didn't sell in unbelievable amounts — that she would rather work with someone else. I have absolutely no animosity whatsoever.

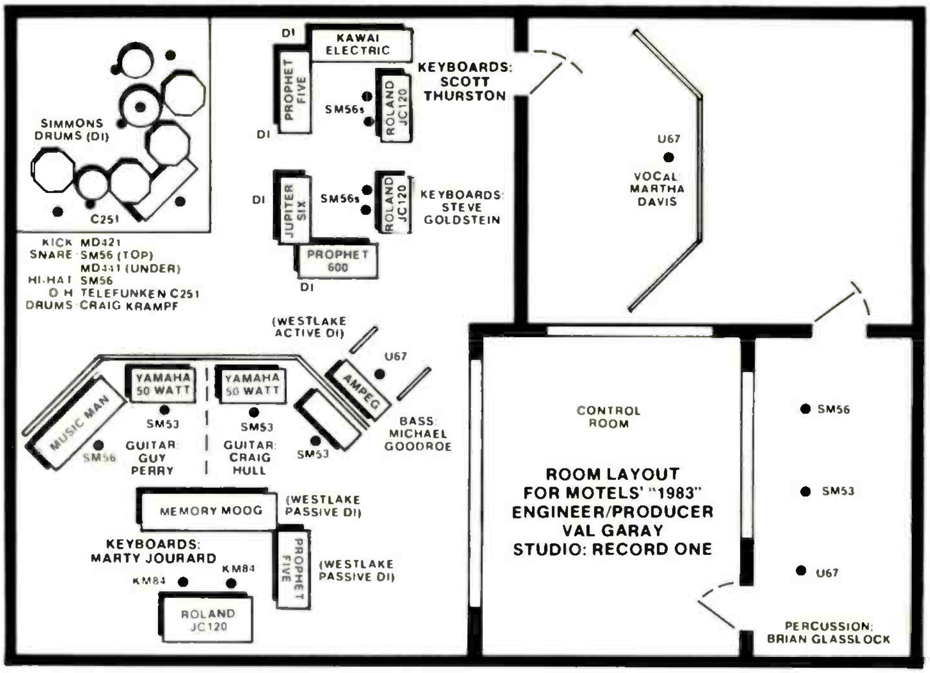
R-e/p: We talked about Kim Carnes' sensitivity before. She seems to have a delicate voice, because of the raspiness. Is that difficult to mike, and get it to cut through the track?
VG: No. If an artist's voice doesn't cut through the track, it's the arrangement that's crowded. That's usually the case.

R-e/p: Is there a procedure you go through for selecting a mike for a particular vocalist?
VG: No. I've used pretty much the same mike for the last 15 years: a Neumann U-67 tube.

R-e/p: Does the U-67 have a special sound for you?
VG: Not really; they're just a great microphone for singers, and I've gotten really good vocal sounds on *all* the vocalists I've ever worked with. You can get things that will sound different — have more edge to it or harshness, or whatever — but you won't get anything that sounds *better*. In some cases I might want something that sounds different than that, and then I've used other microphones. But for the most part, I stick with the Neumann.

R-e/p: So you're going for the accurate representation of the source?
VG: Usually. I use hardly any EQ at all on the vocalist. I must be getting pretty close to the way they sound, because they've never complained that it didn't sound like them when they heard the record!

R-e/p: Which brings up an interesting point about Joan Armatrading. Her voice has a cutting edge to it. Is that the quality of her voice, or was the mike chosen to enhance that edge?
VG: I only did two songs. I used a U-67 for those two tracks. There was no attempt to tone down her voice, or make it more cutting. [Steve Lillywhite produced the remaining nine songs on *The Key*.]



ROOM LAYOUT FOR MOTELS' "1983" ENGINEER/PRODUCER VAL GARAY STUDIO: RECORD ONE

R-e/p: Why did you recut those two tracks?
VG: I didn't recut them; they didn't exist before I did them. It was like her sixth or seventh album, and A&M felt there wasn't a single on it for the United States. They approached me and asked if I would be interested in cutting a couple of tracks with her for the purpose of making a more commercial release for America. I had enough time to do *just* a couple of tracks, so I said, "Sure." She's really a fabulously good artist.

R-e/p: Did you pick those two songs, or were they already worked out?
VG: She played me three or four tunes, and we picked those two ["What Do Boys Dream," and "(I Love It When You) Call Me Names"]. We worked on the arrangements a lot. She wasn't used to that. The person she worked with before, Steve Lillywhite, was terribly uninvolved in the musical aspect of her record. He didn't discuss arrangement changes, key changes, or bar changes. I was a musician long before anything else, so I have to be involved in that. I have opinions and feelings; you don't have to use them, or listen to them. But to not allow me to say them is sort of cheating oneself, because I have good ideas. Obviously, that's been proven. Peter [Asher] listened to my ideas for enough years, so I figure if he's as smart as he is, somebody else should listen, too!

R-e/p: Speaking of Peter Asher, he brought you Linda's last album to mix, didn't he?
VG: No. Not actually, I was contracted to do that album based on the kind of deals we made in years previous. I started recording that album about two years ago. We cut five or six tunes. Then I got in the middle of another album — I can't remember who it was at the time —and Linda got into the Broadway play [*Pirates of Penzance*], then into the movie. Before we knew it, a year had gone by. At that point, I was unavailable, and they needed to finish the album. So we all talked about it when they got back to LA. They came up with the idea of doing it with [engineer] George [Massenburg], who is a very close friend of mine, and a marvelous engineer.

R-e/p: So you were familiar with the album when the time came for you to mix it?
VG: No. They spent another seven or eight months recording more material and, out of the five or six tracks that I recorded, I think they kept three. When it came time to mix the record, George, having worked with Earth, Wind and Fire for all those years, had his style of mixing with those people, and Peter and Linda had gotten very used to *my* style of mixing.

They started mixing with George, and weren't happy with the results — I believe based mainly on the fact that Peter liked my style of mixing. Not because I'm a *better* mixer, because I think George is every bit as good as I am as a mixer. They then approached me on the basis of: "We're old friends; would you do us a favor?" I was right in the middle of another project. "Just give us five days of your time, and try to mix some of this album for us." So I said, "Sure."

I mixed about five or six tracks, and they played them for George so he could get his bearings, because I mixed some of the things that he'd recorded. Now, when you're a good engineer, you hear things — balance, levels, EQ, etc. — a certain way. And when somebody else

. . . continued on opposite page —

changes that, it's *instantly* apparent what they've changed. So, when I mixed a couple of his tunes he became aware of what Peter and Linda were looking for, and remixed again the tracks that I had mixed. The tracks were even more to their [Peter and Linda's] liking. George ended up mixing better than half the album, and I did the rest.

R-e/p: Can you define what was different about your two mixing styles?

VG: Of his initial mixes that I heard, I used more vocal and drums than [George] did. The rest of it is all subtleties. But when you get somebody as good as George is, the subtleties are equally good either way. Do you like Chocolate or Vanilla ice cream? They're both ice cream; it's that sort of thing.

R-e/p: Just one final fact that I was curious about. Capitol chose the Motels' last album, All Four One, *as the first cassette tape to release using its XDR system, which is supposed to improve the quality of pre-recorded cassettes. What do you think of the system?*

VG: I think cassettes are virtually headed for the land of doom, and I'm glad. I think the next realistic avenue is the Compact Disc.

R-e/p: But you can't record on CD.

VG: That's what is realistic about it. Piracy is the *only* problem we have, and it drives me crazy. I understand on one level, and I don't on another. My 14- and 15-year-old nephews — my sister has eight children — were over at my house one day, and we're talking about music. They asked me if I like so and so, and I go, "Yeah, yeah, yeah." And Kenny, the next to the oldest says, "Well, I always go over to my friend's house and tape the albums." And I said, "Don't you know that's piracy. You're stealing from me, your Uncle, who you love dearly." Copy a tape, Go to Prison!

And they go, "Yeah, but the quality of the cassettes in the stores is terrible." And they're right. The cassettes are *horrible*. Because they are high-speed duplicated, the reproduction is [lousy] —the top-end disappears; the transparency disappears. But the tape duplicators have no choice; that's the only way they can make them. If they made one-to-one copies at normal speed, they'd be there forever, and have to charge $20 to $40 a cassette. It's unfortunate, but it's true.

And we [the people who derive their living from records] are losing billions of dollars a year to illegal taping. The Compact Disc will eliminate that. The reproduction is phenomenal. It's small, easy to use; you can drive over the disk with your car; punch a hole in it up to 1mm and it still plays fine, because it's a laser disk. It's almost completely idiot proof.

Albums will eventually become Compact Discs, because the vinyl disk, as we know it, is an antiquated piece of junk. They were designed to operate at 78 RPM. You have to deal with warped records, groove noise, dust, needles. The Compact Disc is the answer. Good-bye to piracy! It will take years, but *that's* where it will be. ■■■

Everything is set. So far, the start of this important tracking session has been going fairly smoothly. All the microphones and headsets are in place, and working properly. The 24-track is aligned, and a set of tones have been printed. Early this morning, maintenance took care of the console distortion reported the previous night.

As the musicians and their equipment start to appear, the guest engineer shows up with his favorite "special" microphones that he always uses on drums. At the same time, a roadie brings in a rack of modified outboard gear that has become essential in producing that famous "hit" drum sound. The assistant engineer quickly replaces the kick and snare mikes, and patches in the customized equalizers and limiters as requested by his client.

The producer arrives and starts getting re-acquainted with the engineer and musicians. After the visiting engineer makes some mike level and equalizer adjustments, everyone is finally ready to record the first song. The assistant rolls tape and, almost immediately, a distortion is heard over the control room speakers. Now a sinking feeling overtakes the second engineer, as he begins to wonder where the source of the problem might be. Could it be those "special" mikes? Maybe it's the modified equalizers and limiters? What if maintenance really didn't fix last night's problem? Or, worse yet, it could be a whole new trouble just beginning.

BASIC MAINTENANCE TECHNIQUES FOR ASSISTANT ENGINEERS

by Roman Olearczuk

Second engineer Dave Dubow/Studio 55, Hollywood

Unfortunately, this little scenario frequently comes in many variations, but there is a logical approach that any assistant engineer can use to isolate and bypass the offending equipment. In this article, basic fault-finding techniques will be explored that, hopefully, will enable the second engineer to improve his confidence in the control room, as well as save the studio costly down time. In addition, this increased knowledge will help him establish a better dialog with the maintenance staff.

In order to quicky spot and bypass suspect equiment, the assistant needs to develop some essential trouble-shooting talents. First, he/she *must* know the proper operation of all the studio equipment in use, including any outside rental gear. Also, signal routing throughout the studio and control room must be permanently stored in their memory for instant access. Next, he/she must increase his/her sensory awareness to observe, feel, smell and hear changes in equipment over periods of time. Finally, a second engineer needs to be conscientious in following-up on all defective equipment.

Trouble reports need to be clearly communicated, either verbally and/or in writing, to technical engineers, so that the symptoms are understood by all involved parties, and repairs can be made easily. Only through a constant dedication towards improvement of these skills will an assistant engineer learn to instinctively hunt out studio problems *before* they become potential disasters to the recording clients. With these points in mind, an examination of fault-finding techniques can now begin.

Studio Equipment

The least complex and, at the same time, the most critical equipment is found in the studio; microphones, headsets, direct boxes, and lots of cables form the basis of all recordings. Quite often it is taken for granted that microphones will work each time they are used. The aware assistant engineer should take note of the condition of each microphone before and after each session's use. For example, any new dents on the outside shell can cause unseen damage to internal parts. The best action is to write up a trouble report by simply stating: "Mike dent on shell; needs to be checked out; please repair by [required date]." With this information, a competent technician can quickly examine the microphone element or capsule for damage, as well as look for

— the author —

Roman Olearczuk recently moved to NBC Studios, Burbank, to pursue a career as a post-production audio mixer. Prior to his recent relocation, Roman was associated with Studio 55 and Rusk Sound Studios, Hollywood. He also serves as consultant technical editor for this magazine.

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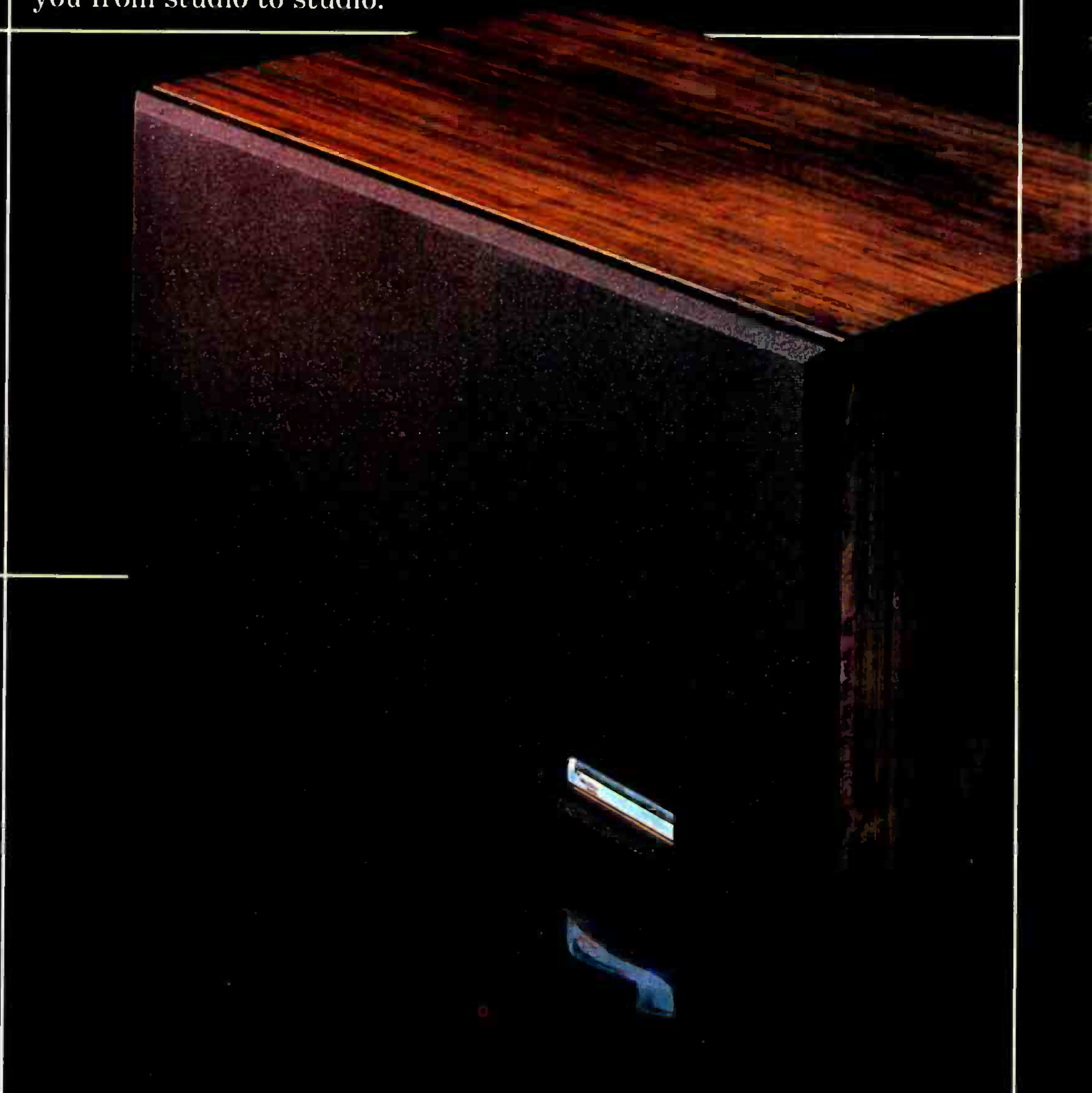

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hairline cracks or breaks within any of the electronic circuits. After some active tests, he can inform the second as to the extent of repairs needed, if any, and whether the device can be back in service by the date specified.

Other physical microphone abnormalities that need to be reported in a similar manner are: rattling internal parts; loose or missing fittings and screws; corrosion on any electronic connection; switches (filter, pad, pattern, etc.) that have become very spongy or very stiff; changes in operating temperature (external/phantom-powered microphones); any excessive moisture, saliva, or liquid spills; and any hint of a carbon-burning smell. Being aware of minute details such as these above will greatly aid in keeping the microphones in top working condition.

As for electronic performance peculiarities, the assistant engineer needs to know first what each type of microphone is *supposed* to sound like when monitored in the control room. By paying special attention to gain level settings when all the systems (i.e., mike, cables, console, etc.) are working properly, the assistant can train his ear to hear the differences between the commonly used dynamic, ribbon, and condenser microphones. He should recognize and look for the following symptoms: increases in noise (i.e., hum, white noise, sputtering or frying sounds); decreases in gain (or no output) within the usual console gain settings; changes in timbre (i.e., dullness or excessive bassiness); and signal break-up with high transient material (assuming the microphone is not being overloaded by sound pressure).

During a session set-up, it is a good idea to check each microphone for these just-mentioned conditions. Tube microphones should be powered on as soon as possible, to give these devices ample time to warm up. Once they have reached their steady operating temperature, any problems with noise, distortion, or low output usually will show up. Also, instead of just "scratch" checking the microphones for output, someone should actually play the individual instruments while the second listens through the appropriate mike channel for any signal distortions (assuming gain and pads have been set properly). The extra effort in hunting for microphone problems *before* the session starts will aid everyone in attaining a trouble-free recording.

Headsets should be inspected with the same scrutiny given to microphones. The assistant should physically look for cuts and nicks in the cable, loose or missing fittings or screws, and cracks in the transducer housings and electrical plugs. Headphones take quite a beating during every session; they constantly are being kicked, dropped, and stepped on, and the cord is always twisted. It is important to look for any signs of physical deterioration before and after each recording date.

Studio 55 second Bobby Gerber check an intermittent fault on a cue box.

The easiest was to test headsets for proper electrical performance is to play dynamic and transient program material (from either disk or tape) through the the cue systems. By boosting the bass on the program source, any hidden transducer distortions will become more pronounced on the individual headset being checked. Basic tracking *always* seems to blow out these marginal earphones anyway, so it's best to simulate the situation before it happens. Also during the listening tests, the headset cables should be twisted over the length of the cord, and given a few gentle tugs at the plug and at the transducer terminations. If any signal disturbance is heard, the headset can be reported and replaced before the session starts.

Most direct boxes in daily studio use are very rugged. However, as stated before, the time to inspect and test these devices is before the session starts. The units should be checked over for any unusual physical and electronic traits. With battery-powered DI boxes, a small note, regarding the date the latest battery was installed, will provide useful information, and can be fastened on the back of the case. If the active boxes don't have battery self-test indicators, perhaps this feature can be added by one of the studio's technicians.

Cables are the lifelines of the recording process, yet these accessories, without a doubt, receive the most daily abuse. Engineers should be on constant alert for any physical signs of damage to the cable: corrosion on the electrical connections; loose shells and receptacles; missing fittings; and stripped or broken threads. Signal continuity should be checked routinely with either a multimeter or cable tester. There are a variety of these sophisticated testers available now, and any one of them would be a useful tool for quickly verifying cable performance. Cables, of course, also can be easily checked while listening tests are being performed on microphones, direct boxes, and headsets. This is an ideal time to give them a twist and a tug. The slightest crackle or buzz is an indication that signal continuity is breaking down.

Having established that the box is not at fault, Gerber moves to the foldback panel.

Frequently-used microphone panels, cue boxes, and other termination devices should be examined for corrosion and loose-fitting connectors. During listening tests, try this trouble-shooting procedure: using a fingertip, simply tap the connector shell. Any distortion, level drop-out, or crackle will signify dirty or possibly worn-out connectors.

As an aid for cable tracing during sessions, studios have been specifying color-coded cables in their new purchases. If this idea is not practical, another solution is to individually number the cables at both ends. Color-coded tape also can be used to signify different lengths. A microphone assignment sheet can be quite useful, not only for set-up, but as a signal tracing aid.

All the techniques mentioned here can be applied to any other equipment used in the studio; the second engineer only needs to apply the same diligence and skill in searching for the potential sources of trouble.

Into the Control Room

Studio control rooms never fail to amaze visitors. To the uninitiated, the huge console, with endless rows of buttons and lights, the programmed tape machines that seem to run themselves, those exotic black boxes that alter sound, and loud music over fabulous speakers, all make the studio seem like the best place to be. But hidden within all that technology are numerous sources of electronic bugs which, at times, give the second engineer a feeling that the control room is the *last* place to be.

Console signal flow should be memorized by the assistant engineer. The function and placement of every button, indicator, knob, fader, and patch point must be known if the second expects to perform his job quickly and correctly. In addition, he/she must master the complete operation and set-up of all the tape machines and signal-processing devices in the control room. Only with this solid knowledge can the second engineer go on to search for and isolate problems.

A majority of the problems encountered in consoles can be attributed to dirt and oxidation build-up on the mechanically-activated components. Switches and pots that aren't used often will crackle under initial activation; the assistant should routinely check and exercise all switches, equalizer pots, send pots, volume controls, and faders for noise. This procedure becomes quite important, especially before tracking dates and large mixdown sessions,

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where every module might be used. Components that become frequently noisy need to be reported and replaced. Patch cords need regular inspection, especially in heavily-polluted cities, where monthly cleaning requirements can be quite common. Frequently used patch points also might require attention for any normal fatigue.

It is a good practice to check gain structure, by patching the console oscillator to each module, and observing the unity gain setings on the fader versus meter readings. Center detent on panpots can be verified in this way, as well as send levels to auxiliary devices. Equalizer boost and cut levels for individual frequencies can be verified. Listening tests can determine any audible harmonic distortion within the complete signal path.

Based on the studio technician's brand suggestion, a can of oxidation cleaner might be a handy control room accessory to quickly cure some of the mysterious noises and signal drop-outs that may come and go. Another useful accessory is a portable vacuum cleaner, similar to those found in data processing centers. This tool will greatly aid in keeping dust, ashes, and other foreign objects from accumulating on and in the console.

Tape machines not only suffer from the same environmental pollutants, but also from oxide build-up over periods of use. Not only should the heads and tape path be cleaned and demagnetized routinely, the area immediately in and around the headstack also should be oxide-free. Use of a vacuum cleaner or air can to get at the dirty areas, if cotton swabs just won't do the job. These particles can easily be kicked up into the tape path when high transport velocities are present, so it's best to keep the area as spotless as possible.

During daily alignments, the assistant engineer needs to listen and observe how the tape machine is reacting. For example, the same alignment should remain stable from day to day. Noticeable level changes might be a clue to problems within individual channels, or even power supplies. Changes in motor noises and bearings might be indications that some component is breaking down. Variance in reaction time from different transport modes can be the start of an upcoming malfunction.

A proficient second should be able to align a multitrack, and check for system drift.

Also, any changes in heat or smell can offer clues to potential malfunctions. The second also needs to know how to detect tape speed changes. When the machine is operating properly, take-up and supply tension should be observed and felt with the forefinger. A known musical piece or tone on tape should be listened to repeatedly on-speed and off-speed, so that when a subtle pitch change does occur, detection will be easily noticeable. It will take time to train one's sensory awareness but, on a regular basis, the assistant engineer will quickly become conscious of extreme — and possibly even small — deviations from a properly operating tape machine.

At the beginning of basic tracking, try this useful suggestion: after the multitrack has been aligned for playback and record, record a tone on the record pad, and individually solo each track for correct playback and sync. Listen for purity of the tone (i.e., no distortion, wow and flutter, etc.). Erase all tracks and listen for clean erasure. If noise reduction is used, try the procedure without noise reduction, and then again with it inserted into the audio chain. These actions will give the engineer confidence that both tape and tape machine are operating properly.

The console patchbay can be a useful place to determine where in the signal path an intermittent or continuous fault occurs.

Outboard equipment comes in a variety of packages and technologies. Processing gear can be tube or solid state, and either analog, digital, or both. The same fault-finding fundamentals apply here as they have been discussed previously. The second engineer needs to be competent in the proper operation of all such devices. Only then can he/she begin to notice changes in signal quality through daily session usage. Fortunately, equipment malfunctions are easier to assess here, since generally these devices are patched either in-line or side-chain. Quite a few of the newer units have bypass switches for instant evaluation of the processed versus unprocessed signal.

The second engineer should carefully evaluate all non-studio equipment as well for proper operation and interface in the control room. As a general practice, all attached cables should be examined for continuity and phase, while connectors should be cleaned and tested. If needed, a technician can quickly evaluate a unit before it is put to use. The studio should have a large assortment of adapter connectors and cables on hand to satisfy different interface requirements. If possible, have the rental companies deliver these in advance of session downbeat, so that ample time is allowed for these evaluations and simple precautions.

Routine maintenance tasks that a second engineer can perform include keeping the multitrack heads and tape path free from dust and oxide residues.

Fault-Finding Techniques

Even with adequate preparation, problems always occur during recording sessions. If all the previous preventative maintenance suggestions have been carried out, then the assistant engineer can assume that genuine equipment malfunctions now do exist. The basic fault-finding techniques to use are: (1) half-splitting, and (2) comparison.

The first method, as its name suggests, splits the audio chain into two parts, and a subsequent listening test confirms which half still has the problem. The remaining portion is split again into two parts and observed. This procedure is repeated until the trouble is found. The second method uses a known, good duplicate audio chain for a subjective comparison against the suspect signal path.

As described, these techniques can be quite useful in the control room. For example, during a mixdown session, a distortion is heard over the left bookshelf speaker. Is the trouble coming from the tape machine, the console, the power amp, or the speaker itself? If the distortion is constant, one can perhaps intuitively rule out the tape machine, but this would still leave three possible sources. Using the half-splitting and comparison techniques, the assistant engineer can cross-patch between the console left feed and the control room right amp. If the distortion moves to the right speaker, the problem is in the tape machine or console. Conversely, if the distortion stays put, then the problem is in the amplifier or speakers. This process would continue until one of these items becomes isolated as the trouble source.

The following studio examples should also provide an illustration of how to apply these techniques to isolate faulty equipment. *. . . continued overleaf —*

Example 1: Basic tracking "buzz" on right-hand main control room speakers.

Equipment in use: microphones, direct boxes, external limiters and equalizers, 24-track, 40 input/32 output in-line monitoring console, audio power amplifiers, and monitor speakers.

Fault-finding tests:

1. Since the input is being monitored, the tape machine is not at fault.

2. Alternate speaker systems are tried with no differences.

Conclusion: Speakers and amps are not at fault.

3. All modules that are assigned with center panning are individually turned off one by one.

Conclusion: Kick, snare, middle tom, and bass inputs are suspect.

4. "Buzz" is narrowed down to bass input module, and its asociated monitor. The first test is to solo the equalizer section. There is no change in the buzz.

Conclusion: Buzz is definitely not in the monitoring, and it's either the microphone, the pre-amp, the external limiter, equalizer, or associated cables and patch cords.

5. The microphone pre-amp output is cross-patched to another module with no change in buzz.

Conclusion: Buzz is coming from the limiter or equalizer, and not from the microphone or pre-amp.

6. Output patch from the limiter is pulled. The buzz level goes down considerably, but there is still a slight amount there. The input patch cord is removed, and the monitors are quiet once again.

Conclusion: Defective limiter, or possibly bad patch cords.

7. A different patch cord is tried from the output of the limiter to the same return point on the patch bay. The buzz is back at normal level.

Conclusion: Defective limiter.

An alternate limiter replaces the defective one. The trouble report is then written up as follows: "Limiter, #1234, puts a buzz on the monitors when its output is patched into a return point on any module."

Example 2: Synthesizer overdub — no signal when playing back a punch-in recording.

Equipment in use: direct boxes, external equalizers and limiters, 24-track with noise reduction, 40 input/32 output in-line monitoring console, audio power amp, and speakers.

Fault-finding tests:

1. The VU meters on the 24-track and console show that the signal is on tape, but somehow disappears on its way to the console.

Conclusion: Source of problem could be either cable connections to the noise-reduction unit, the noise-reduction unit itself, the cable interface to the console, or the console line return itself.

2. The noise-reduction channel associated with this track is bypassed. The encoded signal appears at the console.

Conclusion: Defective noise reduction card.

A spare card is put in place of this faulty card. The recorded passage is listened to, and sounds fine. The trouble report is written up as follows: "Noise reduction card #1234 exhibits not output in playback. Signal OK in bypass and record."

Example 3: Intermittent noise during mixdown.

Equipment in use: External gates, limiters, equalizers, delays, pitch changers, echo plates, special effect audio processors, 24-track with noise reduction, a pair of two-track recorders, 40 input/32 output console, power amps, and speakers.

Fault-finding tests:

1. Since the noise is intermittent, it can be coming from any source in the audio chain being monitored.

Conclusion: Suspect everything.

2. A listening test determines whether the disturbance is truly random, or occurs only when certain musical passages play.

Conclusion: Random noise.

3. An observation of where the noise is occurring in the stereo image is made.

Conclusion: Noise occurs in center over both speakers.

4. All modules that are panned to the left or to the right are muted. (Note: on some consoles, the mute circuitry occurs before the stereo pan-on switch. In that case, the final audio stage within the individual module must be turned off.)

Conclusion: Suspect kick, snare, middle tom, bass, lead vocal, lead guitar, delay, echo channels, along with any external equipment and patch cords, or the quad master and associated circuitry.

5. Whenever the noise appears again, the remaining modules are muted one by one.

Conclusion: Noise is coming from the delay module. However, the source of the noise could be a send to the delay (from one of the other modules), the delay unit itself, or the module being used as the delay return. Also, the associated patch cords could be suspect.

6. The patch from the delay unit is pulled out. The noise stops.

Conclusion: The source is either a send circuit or the delay unit.

7. The return patch is replaced and the noise starts up again. The patch *to* the delay unit is removed, and the noise stops.

Conclusion: The noise is coming from an individual send circuit on a module, or perhaps it is the master send circuit.

8. A pencil eraser tip is used to softly press the associated send pot and its surrounding area on the lead vocal module. It has been determined that this send circuit is the only one feeding the delay unit. The noise increases in level and occurrence whenever pressure is applied near the send level pot.

Conclusion: Noise source is coming from send circuit on the offending module. The lead vocal send to the delay unit is reassigned to another unused send circuit. The trouble report is writen up as follows: "Intermittent noisy send pot #X, or circuit on module XY. Pressing console faceplate causes noise to steadily increase."

Photography by Mel Lambert

Lightly tapping around pots and switches can help track down mysterious faults.

In all three examples, a logical process was followed to eventually isolate the source of each problem. Each test was done one at a time, so that the unique result is based on only one action. This method eventually leads to one single conclusion. The trouble report reflects what was determined during these tests, and is written as a simple, effective statement.

To aid the assistant engineer further in developing an intuitive sense of troubleshooting while he/she becomes experienced in these basic techniques, a table of symptoms and possible faults has been included in a sidebar to this article.

* * *

The information presented here has been of a general nature and is definitely not all inclusive. Specific guidelines for fault diagnosis in console automation systems, as well as other complex and exotic equipment, can be accessed in manufacturer's manuals, or by consulting with the studio maintenance staff. It is a good idea for an inquisitive second engineer to establish a continuing dialog with a technical engineer, a relationship that will benefit all involved. Not only will the second engineer keep track of progress on repair work, he/she will absorb useful pieces of information drawn from the technician's experiences. The increased knowledge gained through a dedication in following these guidelines will enable the assistant engineer to increase self-confidence, and reduce stress in his chosen profession. ■■■

COMMON FAULT SYMPTOMS AND POSSIBLE CAUSES

Observed Symptoms:

No output from mike to tape machine input

Possible Cause:

- Defective or incorrectly placed mike cable.
- Defective or incorrectly placed patch cord.
- Defective microphone.

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• External microphone power supply (if used) defective, or not powered on — check AC cord, fuse, mike power cable, and connectors.
• Console phantom power off, or defective (if used) — check fuse and power-on indicators.
• Mike/Line switch not engaged correctly, or defective.
• Console not set in record or input mode — check individual modules for similar conditions.
• Mike pre-amp gain set too low — check pads and level; possible circuit defect.
• HI/LO variable frequency filter (if used) is engaged, and set for maximum cutoff frequencies.
• Externally in-line patched equipment either not powered, or faulty, or incorrectly patched — check cables for proper operation, phase, and send/return patching.
• Input fader off, or defective circuitry — check for engaged mute and cut switches.
• Unused module is soloed — check solo indicators.
• Automation switches (if equipped) left in read mode from previous mix session.
• Signal not assigned to tape machine, or defective switching circuits.
• Input faders assigned to groups with sub-group master off.
• Input master fader off, or defective.
• Output master fader off, or defective.
• Tape machine not in input mode, or defective switching circuitry.
• Noise reduction (if used) not in input/record mode.
• Cable input to tape machine defective, or not inserted properly — check especially after machine has been moved or worked on.

Low-level input from mike
• Signal padded too much on mike or pre-amp.
• Gain level set incorrectly, or defective pre-amp circuitry — cross patch another pre-amp to check circuits.
• Defective cable — broken signal conductor.
• Defective power supply (if used) on microphone.
• Defective microphone.
• Dirty or defective switches — check all switches on console in-line with signal path (i.e., mike/line, pad, EQ-in, etc.).
• EQ (if used) set incorrectly, or defective — check mike without EQ.
• Defective or dirty patch cord.

Input from mike is dull or bass heavy
• Mike defective or capsule has excessive moisture, dirt, etc.
• Mike impedance terminations to console incorrect.
• Dirty signal routing to console — try different panel XLR and different cable(s).
• Power supply (if used) defective.
• Microphone positioned incorrectly on instrument or voice to be recorded.
• EQ is engaged or defective — check mike without EQ.
• Pre-amp circuit defective — cross patch another pre-amp to check circuit.

Input signal from mike "crackles"
• Defective cable(s) on mike or power supply.
• Defective patch cord(s) or patch points on mike cross patch, or any external equipment patching.
• Dirty or loose XLR fittings on mike panel or microphone.

Mike input signal "buzz"
• Defective mike cable — broken shield.
• Defective mike power supply.
• Defective patch cord.
• Defective externally patched equipment or ground loop — check cabling.

DI input signal "buzz"
• Ground loop to console through musical instruments — try ground lift on DI box, or AC isolation transformer on instrument.
• Defective mike cable — broken shield or improper LO/Shield wiring for unbalanced/balanced terminations.
• Defective musical instrument(s) — isolate or remove individual add-on devices (i.e., phasers, flangers, etc.).
• Single-coil pickup on guitar — try different physical locations of guitar relative to amplifier.

No output from tape machine to console
• Tape machine incorrectly switched to input, or defective circuit.
• Defective reproduce/sync channel on machine — check VU meters on recorder for any indication.
• Console not in line-return mode.
• Noise reduction (if used) not switched to playback mode, or defective circuit.
• Incorrect or defective patch in a return patch point.
• Cable from tape machine to console defective, or not inserted properly to machine.
• Externally in-line patched equipment either not powered, or faulty, or incorrectly patched.
• HI/LO variable frequency filter (if used) is engaged, and set for maximum cut-off frequencies.
• Return fader/monitor pot is off, or defective circuitry — check mutes, cuts, and subgroups.
• Unused module is incorrectly soloed, or defective circuitry.
• Automation switches (if equipped) in read mode.

Tape machine output low
• Off-tape signal recorded low — check playback alignment against actual signal VU meter readings.
• Defective cable from machine to console.
• Noise reduction (if used) defective or not properly aligned — check for dirty switches, dirty level-set pots, and loose card seating.
• Dirty patch cord or patch point on cross patched tape output/console returns.
• Oxide or foreign material (i.e., grease pencil, etc.) on head stacks.
• Tape is shedding, or has excessive drop-outs; check tape path for any build-up of oxide residue.

Tape machine output distorted
• Alignment incorrect — observe VU meters for levels, and check record/playback alignment, including proper bias adjustment.
• Defective record/playback circuitry — check for dirty level pots, switches, and poor card seating.
• Defective cabling to console.
• Output from machine incorrectly cross-patched or terminated.

Tape play speed incorrect
• Varispeed engaged, or wrong speed selected.
• Tape tension incorrect — check against known reference (i.e., alignment tape with frequency counter, etc.).
• Capstan puck very dirty. ■■■

HIGH-QUALITY AUDIO FOR VIDEO POST-PRODUCTION

Computer Control and Synchronization at Regent Sound Studios

by Paul D. Lehrman

Photography by Ron Benzenisti

The magic word "video" is on the minds of many studio owners, producers, and engineers these days, as promises of the media prophets of the past decade are beginning to take solid form. What with cable, subscription TV, direct-broadcast satellite, and all the other growing number of outlets, there seems to be a huge market for new video material out there, and therefore money to be made from this blossoming industry.

But a majority of cable systems in the US still deliver only 12 channels, or fewer, which means that a lot of so-called national services, in reality, have pretty limited audiences. Limited audiences mean limited funds; of the dozens of cable networks on the air, only a couple are actually showing black ink, and a couple of seemingly well-financed cable and STV operations already have closed up shop.

Nonetheless, there are bright spots. A few of the cable companies, like HBO and MTV, are putting significant amounts of effort and money into producing high-quality music-videos. Videodisk players finally are beginning to gather momentum, especially in the area of industrial and educational presentations. Probably most importantly, as film producers and theater owners finally become aware of the commercial benefits of high-quality sound, there is more of a demand for up-to-date audio recording and processing techniques in that medium. [A good example of which is Steven Barnett's articles on the Stones' movie, *Let's Spend the Night Together*, to be found in the December 1982 and February 1983 issues of *R-e/p — Ed.*]

But for a recording studio currently specializing in audio-only sessions to become involved with video means a lot more than just buying a U-Matic video deck and a timecode reader, and sitting back and waiting for the high-powered accounts to come knocking at their door. As Bob Liftin, president of New York City's Regent Sound, puts it, "Anyone who wants to get into this business should get into it with his clients. You'd be crazy to jump into the pot by yourself."

Liftin knows whereof he speaks: he celebrated his 25th anniversary in the recording business last March by buying the facilities of Soundmixers, the six-year-old, four-room audio recording complex in New York's historic Brill Building, and starting the extensive renovations that would turn it into a first-class audio-for-video facility. But, even though he was one of the pioneers in developing modern audio techniques for television and film production, Liftin's involvement in the visual aspects of recording did not occur overnight.

Video Does Need Audio . . .

Regent Sound's original premises on 56th Street was considered by many to be one of the most successful studios during New York's heyday in the record industry during the late Fifties, and didn't begin to service film and television clients until the late Sixties. "Mostly specials where we would prerecord the music," Liften recalls, "which they would then play back on the set and add a live vocal."

Right from the start, Liftin realized that there was a vast amount of room for improvement in the quality of television sound. "The whole thing would be recorded on videotape and then edited," he says. "The problem was that the video editing process was all reel-to-reel, so by the time the audio went out over the air it was five or six generations down. And to top it off, the frequency response for coast-to-coast networks was only good to 5 or 6 kHz."

One of the changes Liftin fought for was diplexed television audio, for which he gives credit to Julius Barnathan, head of engineering at ABC.

"Some people would say, 'What difference does it make? It's all coming over a four-inch speaker,' " he offers. "I never took that attitude. The whole thrust of what we were doing was that *someday* high-quality audio for television was going to be available. If we maintained a high standard, then by the time the technology of the receiver matched that of the transmission, the two would marry together. I have reams of surveys done that concluded that the average public didn't care about good sound, by my record experience taught me that the consumer could *only* judge it properly if the software [programming] was already there."

Liftin found a strong ally for his position in MTV, today a major client. "The original MTV software was promotional stuff," he says, "and the sound was terrible. Half of it was off the [original] records. But what they wanted to do was get a first-generation Dolby [encoded] tape from the record company, and lay that on the one-inch video master, so that what went out over the air was second- or, at worst, third-generation audio. They knew that on the right receiver, it would even be better quality than the record — which has to go through several generations in the pressing, and also has pops and crackles."

. . . and Audio Needs Video

Although the quest for good audio for video has been a major motivating fac-

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tor in Regent's growth, it was a more mundane problem which, in 1974, made Liftin realize that a heavy commitment to video was inevitable.

"We were doing Peter Allen's first album for A&M," he recalls. "Jerry Moss and I were listening to playback, and he said to me, 'Do you have any idea what the studio bill on this album is?' and I said, 'Very frankly, no.' It turned out it had climbed to about $80 or $90 thousand, which we both agreed was outrageous.

"And it wasn't that our rates were high — it's just that so much time was being spent recording this [project] because of multitrack. At the time, we already had an EECO synchronizer for locking two machines together, which meant even *more* time spent on the album.

"I realized that we had completely lost contact with the reality of the economics of making records. We had gone from making albums with a total cost, including the studio bill, of $5,000, to spending $100,000 just on studio time, not to mention musicians' and producers' fees, and whatever else on top of that — and we weren't making bundles of money, either. There was something radically wrong. If records were going to cost this much, eventually it would mean economic disaster for the industry. So I started to look around at ways to use the synchronization technology that we had for *other* media, where it might be more economical."

Developing the Technology

Because of its high price tag, Liftin reasoned, the new technology was more suited to television and motion pictures.

— Regent Sound's Bob Liftin —

But, at the time, techniques for synchronizing audio and video together were at a relatively primitive stage.

"SMPTE code was originally adopted to allow video editors to electronically mark a piece of tape so they could find it quickly," he explains. "We were used to taking two audio reels with the same start marks and code numbers, and locking them up. But a taped TV show, with a prerecorded music track, was a series of short shots done at different times, and edited together in no particular timecode sequence.

"Ideally, we wanted to be able to go back to the original music, marry it to the vocal track from the camera master, and dump it onto the edited videotape. But if the first 10 bars were from a take Tuesday morning, and the next eight from a take Thursday afternoon, and so on, the numbers were constantly changing. So we needed a way to number-crunch the different pieces of code."

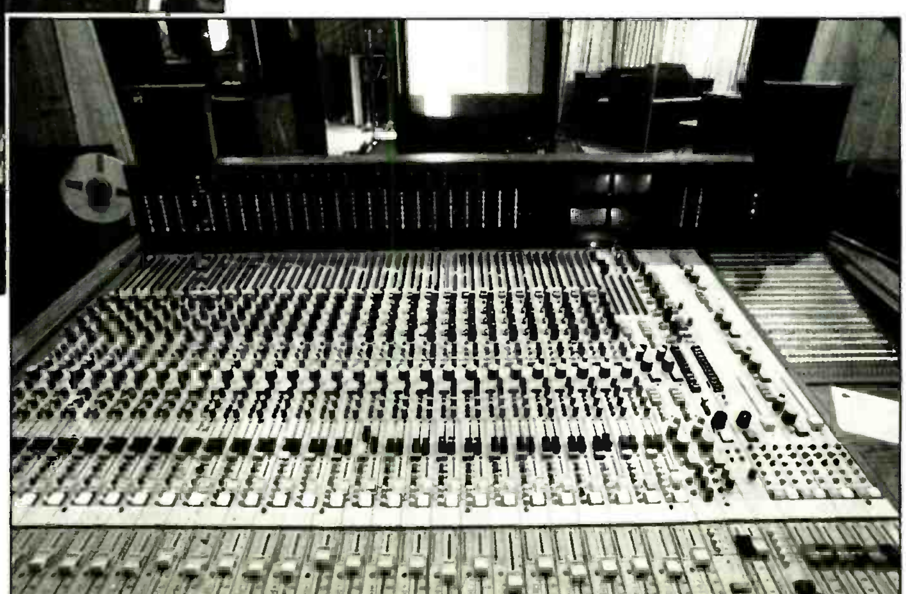
The answer, of course, was computers. Starting with individual microprocessors, in large part co-developed with George Swetland at EECO, Regent Sound designed systems for electronic audio editing, allowing soundtracks to be precisely conformed to video, and then laid back, film-style, from the original tapes on to the finished master reel.

"Typically, the audio was only third generation," says Liftin, "and on some shows, where the video had gone through seven or eight generations, it was *better* quality than the video."

The new technology also put Regent Sound into a unique position in the film world — their ability to manipulate all sorts of timing signals in the audio domain allowed then to salvage projects others considered hopeless. Helping to get things started was the acquisition of a timing generator with exquisite accuracy. "It was developed by Scientific Leasing for the government," Liftin explains. "It was a synchronizer, but it was also a synthesizer, that was good to four decimal places, covering DC to light, at any temperature. I don't know what it was for — somebody like Grumman must have been using it to test something. We leased it at first, because we couldn't buy it, and then we had a better frequency standard than Con Ed or anybody else."

The machine was first brought in for an opera shot in England, with the performers lip-syncing. The audio engineer on the shoot had forgotten to put in one plug on his Nagra, and so the machine had been running free. When it came back to the US, there was no way to edit it, because the timing pulses were all off.

"The Chubb Group of Insurance Companies was liable for something like a million dollars if a technical failure caused a film company to throw a whole project down the tube," says Liftin. "So they hired us to resync the entire track. It came in at less than $10,000. I don't know how much we saved them, but the premiums were something like



Previous page: Studio D, a post-production suite equipped with an EELA console, Ampex ATR-116 multitrack, and two BTX Shadows hooked up to an Apple II as system controller. **Above:** Studio A, equipped with an MCI JH-528 console, Ampex VPR-2B videodeck, EECO MQS-103 synchronizer, and an Apple II for generating audio edit decision lists. **Right:** Studio B's MCI JH-536 console, linked to an Ampex MM-1200 via an EECO MQS-103.

100 grand, so they were way ahead of the game.

"After that lots of jobs came in, mostly from insurance companies. We had one guy send us a picture where the sync signal had been generated by the camera, but it had been stopping and starting one frame at a time, while the Nagra kept running. We not only resynced it, we even changed the pitch of the voices with an [Eventide] Harmonizer. The insurance company called us and said, 'We're off the hook, but they still have to reshoot it — even though his lips are in perfect sync, the guy is moving too fast!' "

At Regent Sound today, computers — mostly Apples ("In 1981, it was the only computer you could get that you could just plug in the wall and go") — handle an astonishing number of functions. They run the mixing consoles, the video editors, and the synchronizers, as well as organizing the tape library, and handling traffic and bookkeeping.

"It's not just bells and whistles — stuff to be fancy," Liftin is quick to point out. "Where it's leading to is reduced costs. Rent and labor and electricity keep going up, and the only place you can cut down is the amount of time it takes to do the recording. How long does a machine take to rewind? Does someone have to watch it? How fast does it cue up and punch?

"Well, we found out about two years ago that almost half the time in the studio was spent rewinding tape. By using wideband amplifiers to read code at high speed instead of tach pulses — and I don't care what the manufacturers tell you, there are always errors with tach pulses — we can cut down on that time."

Computer-Control Applications

Punch-ins and -outs, for example, are considerably faster and more accurate when they are handled by a computer. "The computer knows how long each of our machines takes to punch," says Liftin. "It knows the gap between the heads; how long it takes the bias cards to achieve 50%; which machines require a cue six frames early or six frames late; or whatever. All the engineer has to do is punch up the cue point on the computer. Rehearse functions are automatic too — we've had rehearse built into our [Ampex] MM-1200s since 1979. We also have delay-record: the drum tracks come in at one point, and to keep the overhang in the studio, the engineer might want the brass to come in just five frames later. No engineer in the world could do a punch *that* fast by hand!"

Another important use for the computers has been in speeding up recording of soundtracks that constantly change rhythm. "You might have a piece that's 16 bars at one click rate," Liftin explains, "then 32 at another, then 16 at another. Under software control, it's simple. The machine clicks to a point, and the band records to there. Then the machine backs up, and counts off however many beats the conductor wants at


Studio A's recording area, facing a live acoustic treatment for string and horn miking.

the new click, and sets it up so the downbeat corresponds to where the first click left off.

"The way it used to be done was in pieces, which were then spliced together, compared against the film, and slipped. Now the conductor does a take, and before he picks up his music and walks into the control room for playback, the machines are all cued with video and a piece of audio tape with two clicks on it, already edited together."

What has made all this possible was an admirable amount of forethought on the part of many of the equipment designers: specifically, the inclusion of RS-232 serial interface ports for external software control on the various pieces of hardware. Given the complexities of the field, however, even that control scheme soon may not be enough. "The problem is that RS-232s are really unbalanced lines," says Liftin, "and so they're only good for short distances. So we're starting to think about RS-422s, which are eight-bit, SMPTE balanced serial interfaces, and about 50-ohm data busses.

"Unless someone builds a 422 card for an Apple — and someone probably has — we may get rid of them and go over to 16-bit machines. This would also allow us greater communications links — right now, if you want to use a device that's located in another room, you either have the host computer call it up through a modem, or else you physically bring it in."

"We're always writing new software," Liftin continues. "We have a full-time software writer on staff, as an arm of the maintenance department, who does nothing else. If an engineer wants a special effect, he tells it to our engineering department. They write the program, boot it up in the control room, and he's got his effect.

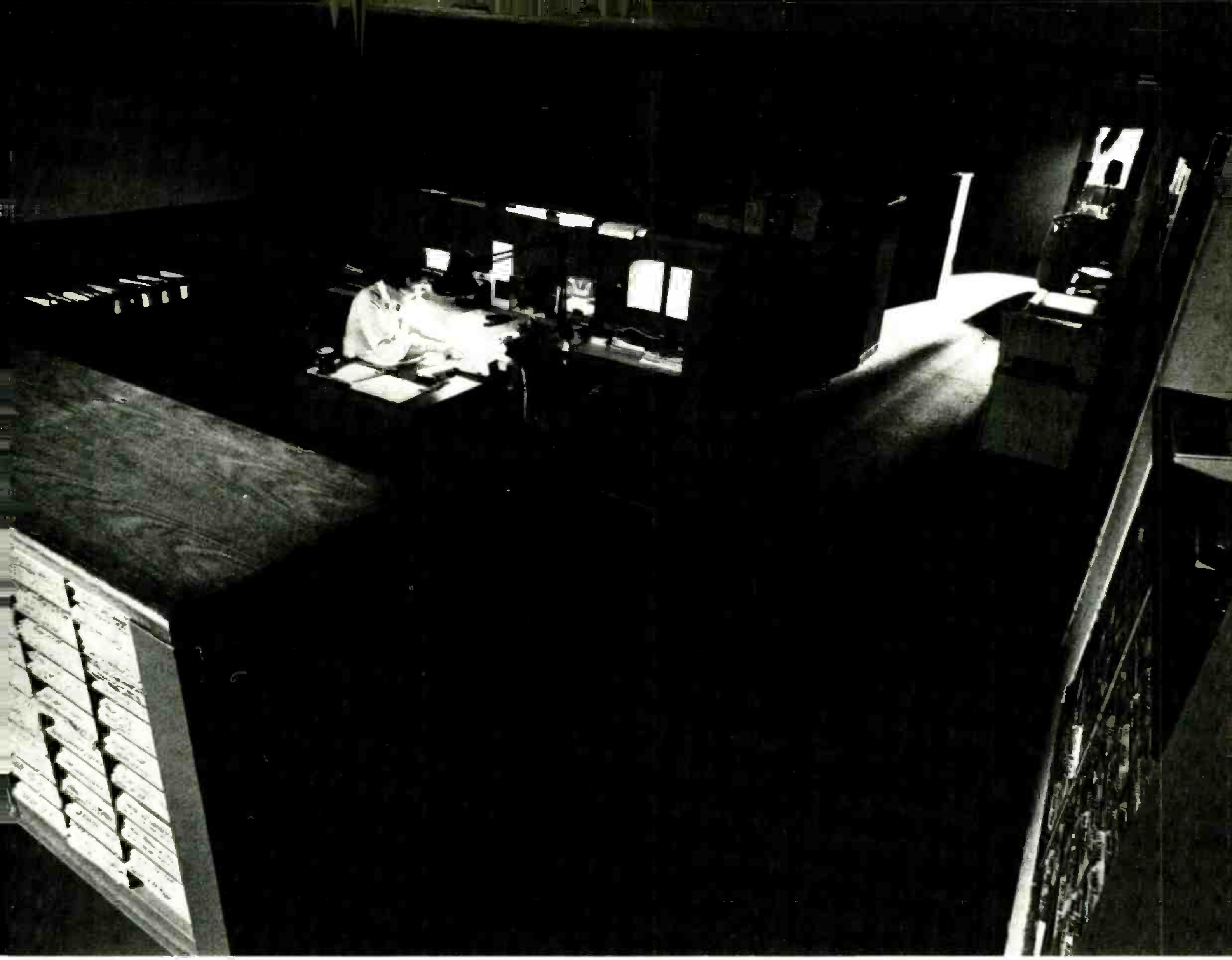
"The ultimate goal is that the equipment become transparent to the engineer. All he should be dealing with is the music. It's [nonsense] that he has to turn and worry and preset and program 4,000 knobs. It should all be automatic."

All of this planning and developing paid off well during the sound editing of the Rolling Stones' concert film, *Let's Spend the Night Together*, directed by Hal Ashby. "We turned it around in three weeks," says Liftin. "Without computers, it would have taken six months."

Of course, being able to perform technical miracles is only half the battle. The rest is developing the proper relationships with clients, both corporate and artistic. For Liftin, an opportunity to considerably improve those skills presented itself in 1976, when he was hired as sound consultant for NBC's *Saturday Night Live*. Actually, in 1975 when the show started, there were two

. . . *continued overleaf* —

Studio C serves as an audio-only overdub/remix room, and features an MCI JH-536 console, shown here linked to a rented 3M Digital Mastering System.



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Saturday Night Lives: NBC's; and one on ABC that starred Howard Cosell. Liftin originally worked on the latter, but when it was cancelled after a couple of months, NBC called him.

"I'm more of a music-production consultant than a technical consultant," he says. "NBC had no experience with rock and roll, so my experience with pop artists and producers from the recording studio end, and my experience with television, let me coordinate between the NBC people and the acts that were coming in. It meant a lot of buffering: you can do this, you can't do that, if you want to do this it means we have to do this. If the band wanted 400 pounds of whatever, I could get it for them, and nobody would get upset. Since I was not an NBC employee, I had a certain freedom their engineering staff didn't.

"It's taught me a lot: where the acts are, what they want and expect — all of which has helped me a lot with specials like Bob Hope, and the Tony Awards. I go in there now and know full well what the performer expects, which makes for a happy relationship. Which, in turn, allows us to get good sound without fighting over it!"

Dealing With Timecode

According to Liftin, the most important aspect to be borne in mind for anyone working with timecode is to ensure that what goes out of the studio is just as readable as what comes in. "People are copying code," he says, "and *not* checking the result. Because they're square waves, it's easy for the pulses to get wrecked going through equipment. By making new code, regeneration avoids some of that, but it also reproduces all of the errors in the old code. The ideal is jam-syncing, but then you need an external clock to hold the old code and the new together. Therefore, there should *never* be an audio or video reel recorded without a 59.94 Hz [video] sync signal on a separate track, so if you ever lose your code, you can go back and rejam the code with the sync signal. Code can get screwed up, but 'hum' is pretty hard to lose!

"Every studio that uses code should at least invest in a reader — even a minimum-quality one — whether they're doing video, or just in-house transfers. If you can read [timecode] with a cheap reader, then you can certainly read it with a good one. If you want to really copy tapes from one machine to another, you have to have a sync generator — a standard clock that relates to all your machines. It doesn't have to be perfect; even if it's wrong, it doen't matter, because at least all your tapes will be related to the same error.

"Standards for interchange between machines are very important. Early on there were problems with some materials that people brought us; some had code on audio track #1, some in the vertical interval, and some on audio #2, which is now the standard for a [video] cassette machine. Now some harebrain has decided to put it on a third track that is in the video. The problem is that no two machines have that head in the same location, and that can cause major problems.

"Having a split-head, two-track reel-to-reel with timecode in the middle is a big advantage [including machines from Nagra and Studer — *Ed.*] For years a big problem was audio machines not being on speed. Now we can not only keep them right on pitch, we can do A/B listening in the studio between the 24-track and the final mix, and always have them locked up perfectly."

Should You Get into Video?

Liftin is very specific about what constitutes the right motivation for an audio studio to invest in video equipment. "If you have a client that wants to do a commercial or something, then do that job with him," he says. "Get the equipment necessary for the job, and feel your way through it. Studios have jumped into audio/video because it's the 'coming revolution,' but that's nonsense. It's more important for a studio to determine what it is, and whom it is going to service. If it's a record studio, it should concentrate on making the best records possible. Nothing beats good recording, whether it accompanies video or not."

And Liftin is insistent about the dangers of studios biting off more than

— continued on page 65 . . .

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. . . continued from page 60 —
they can chew. "The movie industry showed years ago that you can't be all things to all people," he says. "There were sound houses that just concentrated on that — they didn't shoot movies, they just edited sound. Trying to do audio and video under the same roof *doesn't* work; I think it's a big mistake for anyone to try. The audio invariably suffers: if a new Quantel [digital video effects processor] or something comes along that costs $135 thousand, you'd buy that way before you'd spend $50 thousand on audio processing gear, and a new audio deck. That's the *reality* of the business!"

For example, although Liftin's new facility is equipped with elaborate toys like 16mm and 35mm Mag stripers, and a trio of one-inch video decks, he knows his own limitations: "When it comes to film for theatrical release, we only do pre-mixing. Our rooms are designed for television and for records — final mixing for film should be done in a room designed for it. Anyone who tries to do anything else is kidding himself.

"Above all, getting into video is a *human* problem, *not* an equipment one; equipment can always be purchased. Besides, someone who jumps in now will find that the [equipment] will be obsolete a year from now. So you should only buy what you need, and prop up your clients. There's not an unlimited amount of money coming from the networks and the cable companies, so you have to keep costs down — not by cutting rates, but by becoming more efficient. The networks are just as vulnerable as the record companies. Just because it's NBC or CBS doesn't mean it's easy pickin's — they have profit-and-loss statements, too. That's what happened in the record industry. The studios made a big mistake: they figured, 'Oh well, that's Columbia, they can afford it.' But now CBS has shut down all its studios. There is a limit."

Even with all the money the facility has invested in the best equipment, it isn't only the biggest clients that come to Regent sound. "The whole idea is cutting costs," Liftin says. "We get plenty of producers doing their first TV shows, because we can do them quickly, and within a budget. It's just like in the early days with publishing demos. Sometimes the demos were used later for masters, sometimes they weren't. But we've always been able to service anyone."

And at the new Regent Sound there is even a good measure of audio-only work going on. One project underway in Studio C (which has yet to undergo the overtime-heavy renovations to make it video-ready) at the time of writing was Simon and Garfunkel's next album. "There's no reason not to do records," Liftin concludes, "because they can use all that technology, too. Of course, it's all been shifted down from audio/video. We can still offer audio at a decent rate — clients only pay for what they use. The difference is that there's no videotape running during the session." ■■■

EDUCATION FOR THE AUDIO/VIDEO MARRIAGE

Harry Hirsch's Involvement with the School for Audio Arts, at New York's Center for Media Arts

Where are the engineers of tomorrow going to come from? The days when a kid could come in off the street into a recording studio and say, "I like music, and I want to work here," are over. It's no longer possible for an engineer to learn the business simply by watching his seniors, while he earns minimum wage going for coffee, or counting mike stands. Today's studios demand an excellent grounding in theory — especially in areas once considered highly esoteric, such as digital logic techniques, automation programming, control room acoustics, and timecode technology.

In addition, as the record industry continues to collapse under its own weight, those who know *only* how to mix music for disks are finding job opportunities dwindling. To get and stay employable, engineers now have to know how to deal with sound for film, video, commercials, and A/V presentations. Those wanting to break into the field for the first time find themselves facing a serious Catch-22: it's impossible to get work without experience and, outside of the studio, it's impossible to get experience.

Harry Hirsch is trying to change all that. This writer has known Hirsch for a long time, dating from his days at Media Sound, through the construction of Soundmixers and Larry Rosen's Review Room, but I've never seen him as excited as he is about his latest project, the School for Audio Arts, a division of the brand-new Center for Media Arts in New York's Garment District.

The Center for Media Arts is a trade school, in the tradition (and a direct descendant) of the RCA Institute Television School, once the only school in the country where a novice could receive any training in television production. Besides the audio division, the Center offers intensive programs in TV production, video electronics and repair, photography, communications management, and advertising art and design. It takes up 10 floors of a 12-story structure on West 26th Street that used to be known as the Fur Auction Building.

Audio Curriculum

Hirsch designed the curriculum for the audio division, and that's what he is most excited about. "There are 700 hours over seven months," he explains, "but what makes it different from all the other schools is that only 220 hours of that is theory — the rest is hands-on practicum."

And it's not just "17 guys watching one guy at a console," which is how he describes another new program at a music school in another city. In the 55 hours of "Edit Lab" units, for example, each of the 15 students in the class has his or her own Otari reel-to-reel, mixing board, and Tapecaster cart machine. Next to the room is a fully-isolated vocal booth. Here, students get practical experience in recording and editing narration and effects, and cutting music tracks to length.

In the "Mixing Lab" units, students get to work from a common 16-track tape, mixing it to two-track on their own Panasonic RAMSA consoles. At the front of the room, the instructor can push a button and listen to any student's mix, and offer individual suggestions to each of them through headphones via the solo circuit on his console.

At the front of the room is an MGA projection TV and a pair of ¾-inch video decks. "In the 'Production Room Techniques' unit, we're using SMPTE technology," Hirsch continues. "We have $30,000 worth of SMPTE equipment in here. We take a videotape with Dexter Gordon on it, and put the sound on a four-track, striping both tapes with timecode. Then we erase the audio on the videocassette. It's gone — how insecure can you get? We put sound effects and narration on the four-track, then lock up the machines and play it back."

Next door is the film-sound room. "We have [film] synchronizers, 16mm and 35mm dubbers, a Moviola, and a Nagra and fishpoles [booms]," says Hirsch. "We teach them how to double punch, and how to transfer, both fullcoat and stripe." In another classroom, an Ampex MM-1000 16-track stands totally stripped, while students pore over the bias cards, tracing voltages and logic paths. One student shows Hirsch a playback head he has just successfully relapped.

Audio and video tie lines go from the mixing lab, eight flights down, to a fully-equipped 24-track studio in the basement of the building, where a fur-storage vault once stood. Students upstairs can watch the studio sessions on the projection TV, at the same time doing a live mix, each on his own console — to two-track, to videocassette, or to a Tascam 85-16 16-track.

In the basement control room, a class discusses the intricacies of digital delay. In the large studio room is a Steinway concert grand piano, donated by CBS when it closed its New York studios, along with a Yamaha drum set donated by the manufacturer. "The manufacturers are very supportive," says Hirsch, "because they want students to become familiar with their products. Shure gave us 24 microphones. They all know we've got a dedicated group of students here who have committed themselves to this business."

The Center has set up other kinds of external industry support. The video division produces some 50 shows a year for Manhattan Cable, and also is working on projects for WNYE-TV, the UHF channel licensed to the city Board of Education. The audio school is becoming involved with these projects, too: many of the shows are actually shot in the basement studio, and where complex multi-track mixes are involved, audio students get to handle them.

Mixing for the Medium

An overriding philosophy of the school is that mixing for film or television should be treated very differently from mixing for records. In a unit called "Mixing the Image," students are taught, says Hirsch, "that there has to be a front-to-back perspective. A film mixer knows that when a person enters the room, it isn't until he comes close to the front of the screen that he's totally on mike. To have the same intensity at the front and the rear is misleading, so we show them how to adjust the levels. When we mix concerts to picture, we pan according to image. It's the first course anywhere that does this."

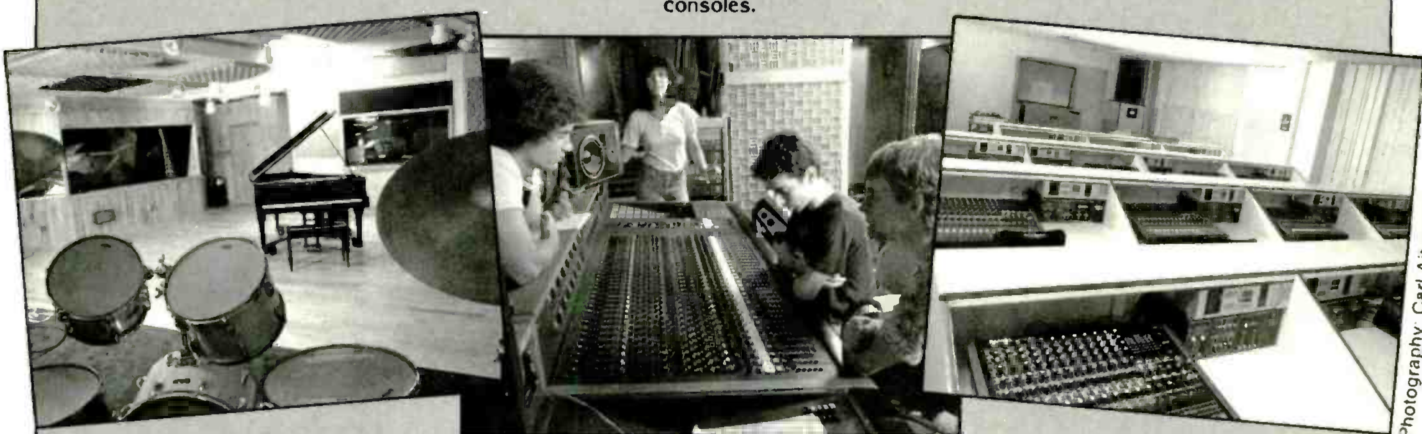
To help students become comfortable with video, there is a unit called "Video Orientation," in which they work with the video division of the school, and participate in shoots. "They get a complete knowledge of the chain of events, and of the terminology," says Hirsch. Of course, it works the other way too — the video school curriculum includes a unit on "Audio Orientation."

All told, there are 24 curriculum units, which also include such skills as studio management; acoustics; location recording (where students go out with multitrack equipment to tape orchestras and chamber groups at the Manhattan School of Music, and the Brooklyn Conservatory); advertising audio; and digital technology. While, as of yet, there's no digital audio equipment at the Center, Hirsch plans to bring in a couple of Compact Disc players, so students at least can hear what the medium sounds like, and he has been promised "a surprise" from a friend at Sony/MCI, "when the clouds clear."

The faculty is carefully picked. "We don't subscribe to the idea that 'Those who can't, teach,' " Hirsch says. "Everyone here is a working professional." Besides the classroom work, students attend a full schedule of guest lectures, given by a corps of experts in the industry, including audio and video producers, equipment designers, musicians, recording company executives, and representatives from such groups as AES, NARAS, and SMPTE.

So far, Hirsch says, the school is doing very well. "Already, we're putting people on a waiting list, and we're not cheap — the course

The fully equipped recording studio **(left)** features a Steinway concert grand, and a Yamaha trap set. Students at work in the 24-track control room **(center)** have access to the latest in processing and recording equipment, and are provided with invaluable "hands-on" experience. In the Mixing Lab **(right)** each student has the chance to mix a common 16-track tape through individual RAMSA consoles.



Photography: Carl Albanese

costs $6,100." There are quite a few foreign students. "Good news travels fast," he smiles. The school is a member of the National Association of Trade and Technical Schools, and is licensed by the state. For needy students, financial aid is available, and the school administers federally-funded Pell grants, and Veterans' educational benefits.

Most of the students are between 18 and 23, although the age range goes as high as 30. "Most of them are looking for entry-level positions," comments Hirsch. "Some of them may know some aspect of the business. We also have night courses for people who have always fantasized about being in audio."

One student this reporter talked to is a fairly successful film and TV composer, who came to the school because he was "tired of being jerked around by recording engineers. I wanted to learn how to do this stuff myself."

Some students come here after they have finished college, while others go from here to a four-year institution. Several of the Center's divisions are involved in a co-operative program with St. John's University, and plans are afoot for the audio school to team up with the music division at New York University.

Emphasis on Practical Training

In an industry which too many people want to get into for romantic reasons — glamour, prestige, and the chance to meet Mick Jagger — Hirsch's curriculum is an oasis of practicality. "The coarse is not only for the recording industry," he points out. "It's for the *audio* industry. There are people who come in here saying, 'The only goal in my life is to mix Gold records.' But later that same guy says to me, 'Harry, I dream to edit. I'm really getting off on it.' And that guy gets a job at the UN. There's more to the world than mixing Platinum. They may find they like SMPTE technology, or image mixing, or location recording, or A/V.

"Graduates can go into CMX editing rooms, video studios, ad agencies, even the networks. When ABC hires for the Olympics, I can have people go out there who have had 160 hours of trouble-shooting. You'll notice I haven't even mentioned the word 'recording studio.'

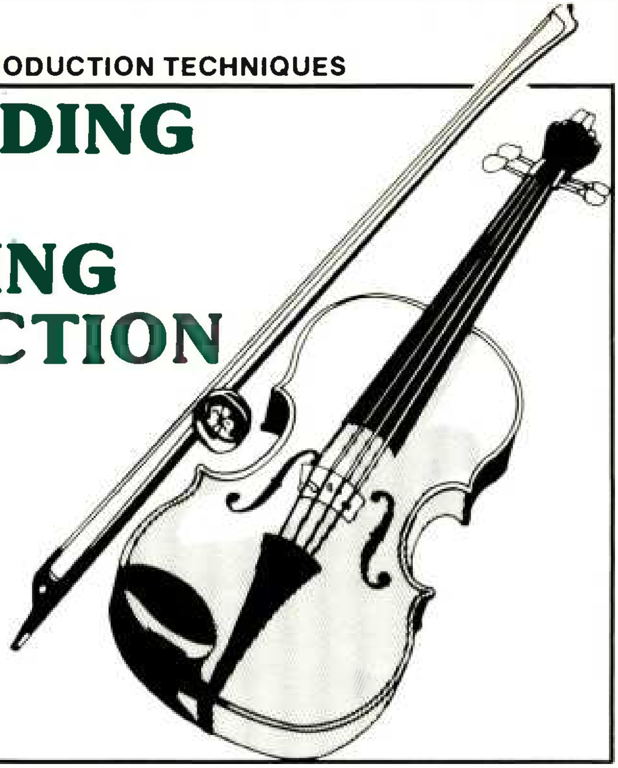
"We have a unit called 'Audio/Visual Production.' We have a library of 1,000 slides of the place, and they pick out a hundred, beep them, write a narration, put effects on, and then at the end of the week call me in for a slide show. Now they've got themselves a business — any of them can go out there and set up shop doing this."

Perhaps the most impressive aspect of the School for Audio Arts is the obligation it feels towards its students in realizing the benefits of their investment. Not surprisingly, the School uses the technology at hand to help its students find work. In a 25-hour unit called "Self-Marketing and Job Search," students go through a mock interview, with a video camera pointing over the interviewer's shoulder.

"The student sometimes comes out of that very disturbed," says Hirsch. " 'That really can't be me!' But the instructor can point to the tape and say, 'Yeah, that's you with the hunched-over shoulders and the apologetic attitude. *I* know you're the best mixer in the class, but you have to show that *you* know it — you have to show a little self-assurance.' We have a responsibility to improve their image, so we take a lot of time with that."

That responsibility continues after the student has graduated — the school offers a lifetime placement service for anyone who finishes any of the programs. Although as of this writing the audio division has yet to graduate its first class, and it is therefore impossible to determine its track record, Hirsch is quite confident about his students' prospects. For one thing, the older divisions fo the school boast a placement rate in excess of 80%. Harry Hirsch expects, with good reason it would seem, his division to do at least as well. □□□

RECORDING THE STRING SECTION



by Robert Carr

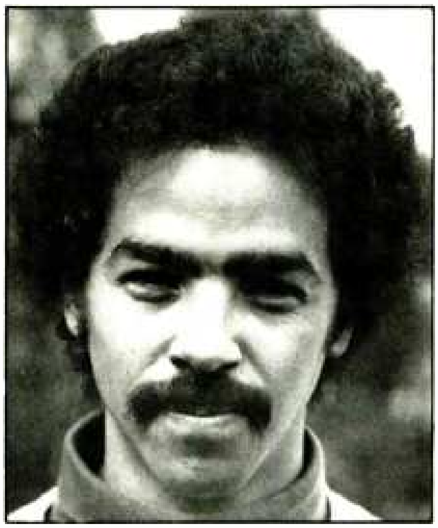
Ever since the advent of electric keyboard instruments, players and design and recording engineers have collaborated to reproduce electronically the sound of a violin. From those earliest attempts came so-called "string machines" that have been described as faithful representations of fingernails scratching a blackboard. (That's one of the nicer critiques.) During the years that followed, of course, sophisticated analog techniques and, most recently, digital sampling methods have afforded musicians the ability to emulate the sounds of entire string sections with uncanny accuracy, for leisurely dubbing on to low-budget, multitrack extravaganzas.

Yet despite all these spectacular advancements, one aspect remains conspicuously absent: the life force or energy or stirring inspiration — call it what you will — that even a mediocre symphonic presentation is capable of generating and transmitting. Sadly, the economics of music production is slowly eliminating the feasibility of hiring and recording large sections of players for album, broadcast, and feature-film projects. Along with this persistent nurturing of "the bottom line" comes, perhaps, the inevitable extinction, or at least diminishing, of the art of orchestral recording.

Ironically, the acceptance of the synthesizer as a monetary savior has, in some cases, backfired.

"There are so many dates where the synth player spends a lot of time in the studio trying to get a sound or figure out a part," says veteran engineer Armin Steiner, who presently handles all TV and film recording at 20th Century Fox studios in Century City, California, and specializes in string sweetening for record dates. "We could actually bring in a great arranger, have him and a small orchestra lay down all the parts in about three hours, and end up with approximately the same costs as using a synthesizer for two days!"

Granted, this may not be a usual situation; most of the time hiring a synthesist saves money. But, when quality is necessary, nothing compares with the textures and versatility of real strings. Television orchestrator/composer Stephen Taylor describes them via an analogy with color. "No matter how you write for strings, and what range you



As a member of the scoring team working with Mike Post and Pete Carpenter, **Stephen Taylor's** musical compositions and orchestrations appear regularly on several major television series, including *Greatest American Hero, Magnum PI, The A Team,* and *Rolling Thunder*, as well as a number of shows not affiliated with Post/Carpenter, such as *The Incredible Hulk, Father Murphey, Faeries,* and *Thundarr*. This Stanford University graduate's piece, *The Sugmad is Dreaming*, was commissioned and recently performed by the Pasadena Chamber Symphony as an overture for its 1983 season.

put them in," he offers, "their sound is a lot more transparent than any other orchestral group, being sort of a pastel kind of charcoal, rather than a bold line in acrylic. You really can't make violins thick like a horn part, even if you write a unison line for 40 violins. You get just a full section sound. The only exception I can think of would be to mike the instrument like [violin virtuoso] Jean Luc Ponty, for a cutting edge and more scratch of the bow."

Taylor goes on to explain that a string accompaniment for a lead vocal written to comprise first and second violins, violas, and celli actually may be quite complex, with each section playing an active moving line, much like a Bach Chorale. "That would look pretty dense on the score. Yet you'd get a very lush, fat string texture where you could pick out each part, and the vocal or lead instrument would stand out against that flowing background. If you tried that type of writing with brass instead of strings, the accompaniment would sound busy and awkward, and probably obscure the lead line."

All of which might explain why live strings appear to be so difficult to blend with the music of a rock group. The loud, driving sounds of electric bass, drums, and guitars are all quite opaque, and tend to mask the thinner, more delicate textures of legitimate acoustic instruments. Armin Steiner has discovered that larger sections are most desirable for a rock or commercial pop date, "because you don't have to use as much to get the impact. A smaller string section yields a thinner sound that doesn't lend itself to 'wrapping around' the rhythm section. The tendency is to boost the level, which doesn't sound big enough to compete. A full orchestra provides that sonority and fullness, so you can back the amplified rhythm section into the track for a better result."

Stephen Taylor offers a suggestion for double tracking that may create the audible illusion of a large section, rather than simply recording the same part twice. For the sake of this example, suppose the session comprises six violins, two violas, and two celli. Record the first pass as written, and pan those to the left. Before doing the second pass, however, change the parts slightly. Don't necessarily change the notes, he stresses, but throw in a grace note [short introductory note before the main note] here and there for the violin; add two or three pick-up notes to a phrase for the viola; or every once in a while write a passing eighth note into a phrase of quarter notes that is a little lower or higher than the rest of the line. A few modifications of this kind throughout the section parts paints the illusion of more string players when panned to the right in the overall track.

"The two parts lock together like two hands might," Taylor says. "For the most part, you hear the doubling as you would if you had done two identical passes. But there is just enough varia-

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FILM SCORING SESSIONS
Danny Wallin's Wide Range of Experience

Drawing on his years of experience in the recording business, concentrating primarily on large ensemble scores for feature films, Danny Wallin helped redesign the Record Plant Studio M scoring stage located on the Paramount lot in Hollywood, California. [See the October 1982 issue of *R-e/p*, "Design and Construction of Film Scoring Stage," page 128, and "Microphone and Recording Techniques for Film Scoring," page 134, for an in-depth coverage on his recording philosophy — *Ed.*] Wallin's list of scoring credits appear to include practically every major motion picture that has hit the silver screen in recent years: *Star Trek II, Rocky III, Annie, Trading Places, Dr. Detroit, Altered States, Finian's Rainbow, Deliverance;* the list goes on and on.

Room Design and Layout

"I had Helmholtz resonators and a tunable louver system [or series of rotatable wooden slats] installed around the room to control the amount of vibrations in certain frequency ranges," Wallin recalls. "What you hear now is really the upper midrange, which I think is the 'speaking range' of most instruments. I was trying to make an environment that sounded live without having [an] excessive reverb time. In this room it's very short — about half a second; that's enough. When you stand in there, it sounds really live. It's just a matter of very little long-term leakage.

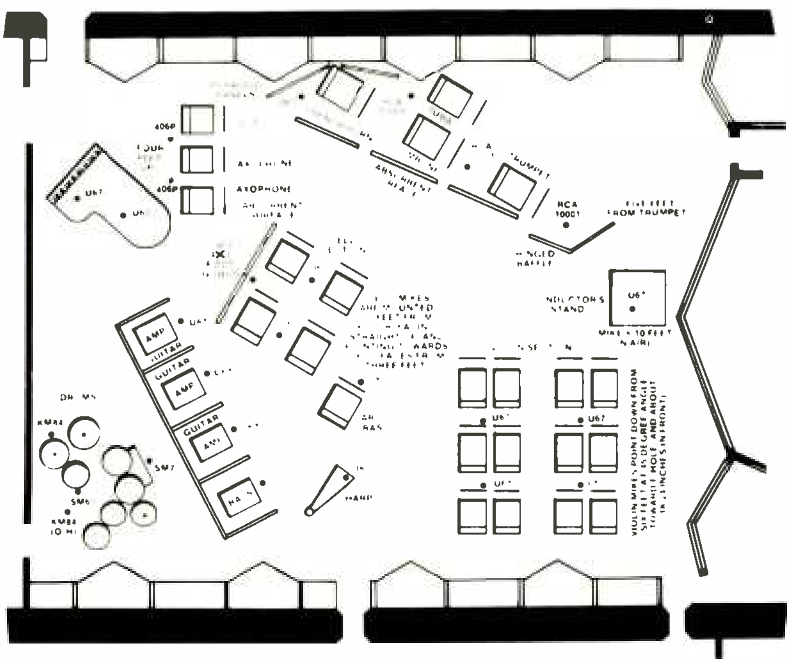
"The players can hear pretty well. Once in a while they'll complain if the distance from side to side is too great. But the way it is tuned now, the room itself is contributing very few secondary reflections. Standing waves are minimal, and you hear a lot of direct sound. The room is great for quick passages, and 'speaks' well for the microphones.

"Because the room has so much presence, it sounds symphonic with just one M-S mike placed above the conductor's head, picking up correctly what he is hearing. But that doesn't give the impression of a 'Hall.' I have acoustic chambers [available] for reverb, but for more live sounds, I've planned another treatment to the room. The upper section of the studio, which is all draped with velour, will house panels that are 2½ feet wide, with small wooden wedges on one side and, on the other, material that is absorptive in the upper midrange and high-end. Those panels will be located on tracks, and rotatable for tuning to provide basically the same flexibility I get with the louvers.

"Then I'll place another ambient mike [a Calrec Soundfield microphone with multiple directional patterns, but left primarily in the M-S mode] way up near the ceiling, and about 15 feet forward of the conductor's head to read just those panels. That, added in with the other ambient mike, should give me the 'Hall' sound. One microphone is reading the actual orchestra, while the other is reading the upper room."

Section Miking

In addition, Wallin mikes all the sections. Although the microphone selections stay relatively the same from one session to another [see diagram below], the mike positions



tion in the melodic lines to sense the difference between the right and left channels."

String Family of Instruments

Several articles ago in this continuing series on miking techniques we focused on acoustic and electric guitar. The original intent then, as it has remained to one degree or another during the course of this series, was to convey an overall view of certain instrument families. Never can it be repeated enough: recording is more than just placing a microphone somewhere, and turning on the tape machine. Sound starts with the instrument itself, and understanding the principles and idiosyncracies of how that musical device operates is one of the keys to unlocking the mysteries of making a good master recording.

In addition, a little knowledge may be invested as a conversation opener with musicians to whom you otherwise might not speak. Those players know how their instruments *should* sound; they're experts and are able to provide you with insights that probably can make your tapes sound better.

For a more in-depth study of stringed instruments, again I recommend the book reprinted by *Scientific American*, entitled "The Physics of Music," and which contains detailed drawings and explanations about the violin (comparisons to the viola, cello and double bass, too), and the concept of vibrating strings in general. For the time being, a few pertinent facts should be noted in order to provide a brief familiarization with this traditional orchestral family.

According to David Rivinus, one of the principals of Metzler and Rivinus Violins in Glendale, California, the basic design of the violin, viola and cello

David Rivinus began his training with a local violin maker in Indianapolis, Indiana, by the name of Thomas Smith. After two years of learning the basics, Rivinus headed for Los Angeles to apprentice with internationally recognized instrument builder Hans Weisshaar, where he spent four years learning his craft. Since then, with a fellow apprentice, Thomas Metzler, Rivinus has opened up his own shop named, appropriately, Metzler and Rivinus, in Glendale, California. Currently, their clients include members of the major symphony orchestras, and recording-session ensembles based in the Los Angeles area.

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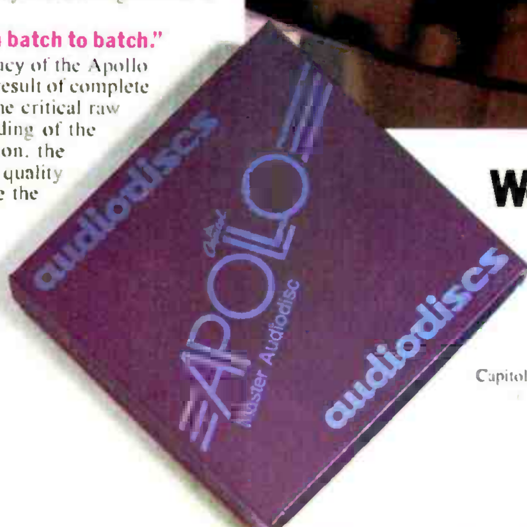

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essentially is the same, with the exception of a few proportional differences. (And a check with technician John Peterson at World of Strings, located in Long Beach, CA, confirmed that the double-bass construction may be included in this comparison.) Because of these similarities, discussing one of the family — i.e. the violin — will provide a suitable reference for grasping the mechanics of the rest.

All four body shapes are natural, efficient amplifiers that radiate the majority of their sound from the f-holes in the front face. Rivinus points out the violin is a relatively small instrument in relation to the amount of sound a well-constructed specimen can produce. (Theoretically, the viola, cello, and bass all should be proportionally larger than their actual sizes to support their relative frequency ranges, according to "The Physics of Music.") Responsible for this phenomenon are two devices incorporated into every stringed instrument: a bass bar; and a sound-post.

The bass bar is a moustache-shaped piece of wood glued to the underside of the violin top, and running lengthwise just below the bridge foot that supports the lowest string. "It's function is to coordinate the vibrations of the belly of the instrument, and to enhance the bass response," says Rivinus. "Without this bar, the low frequencies would be thin, soft, and airy."

The sound post, essentially a spruce dowel, is friction-fitted between the top and bottom faces of the instrument, and sits just behind the bridge foot that supports the highest string. Because the dowel is literally wedged between the two faces, a violin maker can reposition it when necessary to adjust the qualities of the instrument. "The post transmits sound from the top to the back plate," Rivinus explains. "The idea is to coordinate the two plates and make the body vibrate as a unit, instead of as a whole bunch of independent membranes. The instrument then puts out the optimum amount of sound."

"A curious phenomenon about the violin is that, close up, it may not sound very loud at all," Rivinus continues. "But a well-constructed instrument appears as loud, and sometimes louder, in the back of a huge auditorium, as it does under the player's ear. That's one of the reasons why many antique instruments, such as those manufactured by Antonio Stradivarius, are so sought after. You may barely hear a lesser-quality violin, but a 'Strad' sounds tremendously loud, even in the back of a huge hall, like the Dorothy Chandler Pavilion [Los Angeles]. There are several clues and theories about why that occurs, but no one really knows for sure."

Although there presently exists no conclusive and tangible parameters that may be copied to reconstruct such a design today, quality is not limited to the hand-mades of yesteryear. Rivinus

— continued on page 76 . . .

FILM SCORING SESSIONS — *continued . . .*

change to eliminate intermodulation problems whenever the sections change. "That's the secret to string miking — eliminating the intermodulation on the high-end of the strings. Those are the beats in the higher frequency range that actually hurt the ears. The miking I've been using has been correct for picking up the sounds, but there was still something that wasn't right. Then I discovered that by moving the string players slightly apart by about 6 inches to 1 foot, I could get rid of the intermodulation, and that solved the problem. [Wallin also is well known for his fondness of tube condenser mikes, including the ubiquitous Neumann U67 — *Ed.*]

"If you notice, there are very few baffles between sections — just a couple of three-footers. Those separate the strings from the brass and woodwinds for dubbing purposes, because we're going direct to [35mm] Mag. But there are no baffles at all among the strings, and it sounds very open. For a classical sound, there are never any baffles in the room at all.

"On those occasions when you want to mix together the classical scoring sound and the up-to-date rhythm/keyboard sound, you have to use some baffles. You can still get the nice open string sound, while you eliminate the cross leakage that is characteristic of the symphonic approach. The brass can creep into the strings a little, but you keep that real clean rhythm sound that everybody likes."

Rock Session versus Film Scores

"I would cut the strings and rhythm section at the same time for a rock date," Wallin offers, "but I wouldn't change my miking technique at all. Basically, I'd keep the same approach that I'd use for a big symphony. That's the best way to record string sections.

"If the players wanted a hotter string sound, I may tighten up the miking. But if you don't have the intermodulation, the sound is really hot already — if you want it to be. Intermodulation means you're getting cancellation, and without that cleanness the overall sound is kind of 'shaggy' and weak, and doesn't cut as much. I don't say the closer miking is wrong, it's just what I like to hear, and what works for me for that hot kind of string sound.

"If you have 30 strings, and you mike them really close, you're hearing individual strings, *not* the harmonics of 30 strings. Then again, 20 to 30 players is a small section. We usually work with 60 to 80 pieces most of the time — 60 strings, 12 woodwinds, so on. For a small string section though, I try to keep the players all to one side of the studio. They can hear each other better that way, rather than being spread from wall to wall.

"Feature films and records are all recorded on 24-track, so we can do whatever we want

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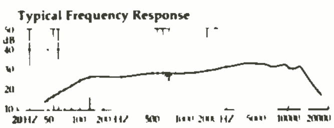

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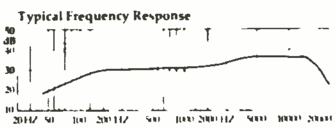

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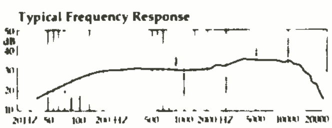

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stresses that "well-made contemporary instruments are also capable of producing the same results."

String Selection

Of primary concern to the recording engineer are those factors which alter the sound of a musical instrument. Of course, the same playing techniques utilized by guitar and electric bass players apply — specifically where on the string the music is played; how and how hard the string is set in motion; and so on. But certainly the *type* of strings employed by the various players yield totally different sets of tonal characteristics. According to David Rivinus, who offers the following encapsulated overview, currently available and in use are three varieties of strings, each composed of a core tightly wrapped with a second material. The earliest strings were made of gut, a product of sheep's intestine, which was carefully treated, worked down to a certain diameter, and stretched on the instrument. Although they were extremely responsive to the player's touch, and produced a sweet sound, they also were unstable in varying environments, and broke easily. Plain gut strings still may be found on occasion, but gut wrapped in silver or aluminum stabilizes the negative reactions, and adds a bit of power, while reducing the responsiveness.

Steel-core strings were introduced in the 19th Century, followed recently by a petroleum distillate-type called Perlon. The steel core surrounded by a second steel alloy (like a piano string) is dependably stable, and produces an extremely loud and powerful sound. The obvious disadvantage is the often harsh and brassy qualities inherent to metal components. For that reason, many professional players choose steel for use only as a high string. The Russian school of playing, however, is the one exception, because they are looking for a more 'aggressive' sound.

The Perlon core, wrapped in silver or aluminum, has a tendency to produce a sound closer to gut, Rivinus concedes, yet without the annoying disadvantage. The sweetness of sound is preferred by most players.

The Concertmaster's Role

The problem of coordinating large numbers of musicians is not new. And, as with most recurring events, necessity has established a hierarchy that at least should be addressed and, for the wise engineer, can be used to advantage. The roles of the composer and conductor are self-explanatory, and need not be considered. The concertmaster's purpose, however, tends to be misunderstood or understated, and such confusion could possibly lead to embarrassing or conflicting situations.

The person sitting in the "first chair," according to Los Angeles concertmaster

FILM SCORING SESSIONS — *continued . . .*

to them later. But in film the final mix usually goes to [35mm] three tracks. The [film mixing] dubbers want all the strings on one [track], brass and woodwinds on another, and the keyboards and percussion in the center. If the picture is [being released] in stereo, then I prepare a stereo mix, just as I would for a record date: strings stereo left and right."

Synthesized and Real Strings

"For a record date, we record the synth and strings at two separate sessions," he explains. "We usually do everything at once for film. We just mix the live and the synthesized parts together for whatever blend we want. No problem.

"We don't do much double tracking, because if you double track at a film date you have to pay the musicians for another session. You might as well bring in the people to do all the parts at once. Record sessions are a different story; double tracking happens all the time. If you have the strings miked with no intermodulation, they sound very big and full. The IM that is normally present won't hurt the other tracks, but within the string tracks themselves the problem might add up. I know that every time you go down another generation, the intermodulation seems to get worse." ■■■

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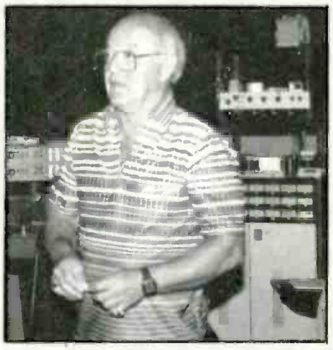
TELEVISION AND FILM DATES
Mickey Crofford's Miking Techniques

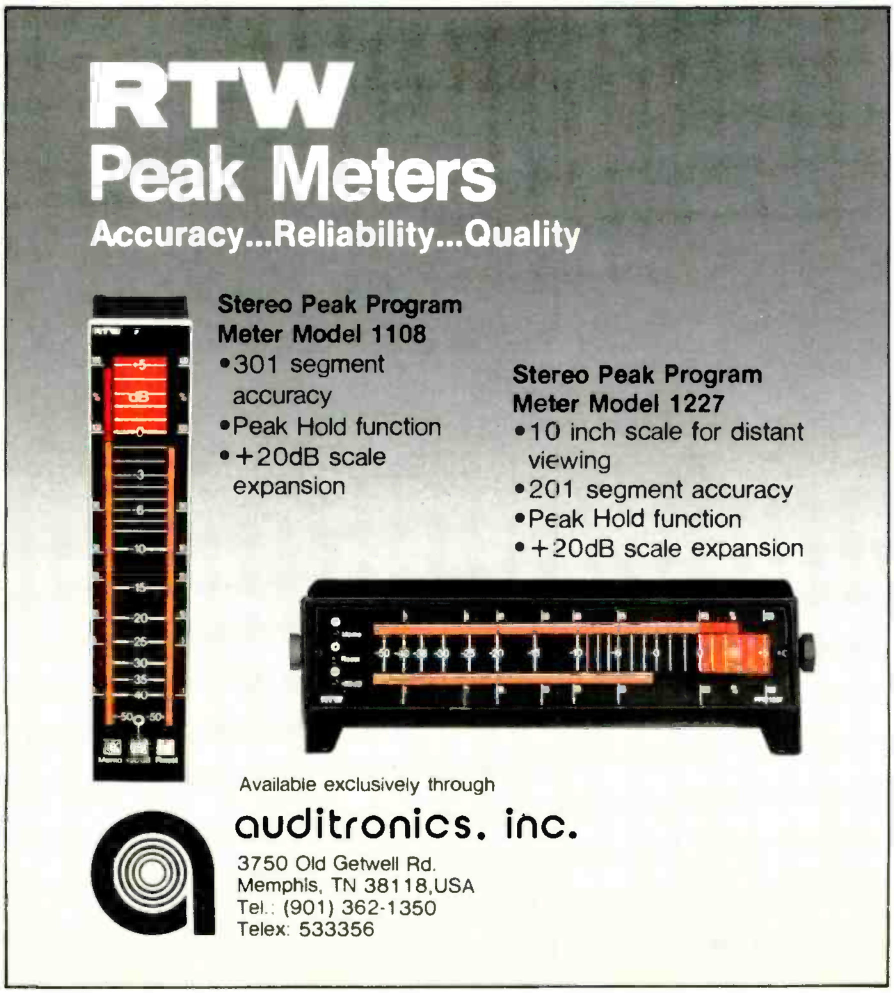


Mickey Crofford started engineering around 1950 in New York, where he worked for Westminster Records and RCA. His move to RCA on the West Coast came about 17 years ago, where he continued with record dates, but soon branched into recording for TV. A stint with John Denver Associates followed, until six years ago when he moved to Universal Studios' Sound Stage 10 in Los Angeles. Since then Crofford has worked primarily on television shows such as *The A Team, Magnum PI, Quincy, Knight Rider*, and *Greatest American Hero*, as well as films like *Pyscho II* and *Masada*.

Room Design

Sound Stage 10 is patterned after the old (now closed down) RCA scoring stage in Hollywood, measuring 25 feet high, 75 feet wide, and 40 feet long, with sound insulation all the way around the perimeter. The temperature is strictly controlled, and remains consistent from one session to the next. Although the dimensions are quite large in terms of recording space, Crofford feels that the room is pretty dead, even though it may not sound that way. "Most of the big orchestral leaders, like Jerry Goldsmith, think it's *too* dead," he

Overhead microphone layout for violin section (left), and cork tiles placed below viola and bass to protect flocr.



Sidney Sharp, functions as liaison between the conductor and the string players. He tries to make the string sections conform to the musical intent of the conductor or composer, by defining fingerings and bowing movements. He conducts the orchestra, sometimes a section at a time, while the conductor evaluates the levels and tones in the control room. And finally, the concertmaster plays any violin solos, should the need to play one arise.

"The soloist is not necessarily isolated from the rest of the orchestra, Sharp says. "Often I remain where I normally would sit, and the rest of the orchestra plays softer. On occasion, I simply stand at the right time and play into a microphone that the engineer has set up for that moment."

"I usually like to go into the control room during playback," he continues. "If the strings don't sound right, I'll make a suggestion — either the mike is too far away, too close, or whatever. That way I can work on a blend for the section, too."

Cueing System

Generally, most string musicians prefer to avoid the use of foldback headphones if at all possible. For a symphonic set-up, where no contemporary rhythm section is present, baffles can be eliminated. The conductor then balances the orchestra within itself, and everyone hears the other sections just as though the session was a live performance.

But you'll find that certain types of string dates dictate the use of headphones by the players. Something like

After graduating from the Curtis Institute of Music in Philadelphia, Pennsylvania, **Sid Sharp** joined the Philadelphia Symphony Orchestra, just long enough to gain sufficient performance experience. Relocating to Hollywood proved to be the right move, because he soon became a member of the LA Philharmonic at the Hollywood Bowl for Stukowski in 1946, and concertmaster in 1947. The next year, Sharp decided to enter the world of commercial recording, and signed a contract with Paramount Studios for a couple of years. Since leaving the studio, he has been an independent contractor acting as concertmaster for the majority of string sessions in that town.

TELEVISION AND FILM DATES — *continued . . .*

says, "but they get used to it. Most of the time they don't have the luxury of casting a room for a particular sound. The film company usually wants them to use the company facility for recording, unless they have a contractual agreement to go where they want, which is very rare."

Because the studio is producing regular series for weekly television broadcast, the composers, conductors, arrangers, and musicians generally stay the same for the entire year. "If the director changes, he usually brings in a new conductor, and the players stay the same every week unless the conductor has the liberty to pick and choose his sections when those people are available. [Players are contracted only on a per-session basis.] That all makes my job easier, because I get to know how everybody plays, and what the people want."

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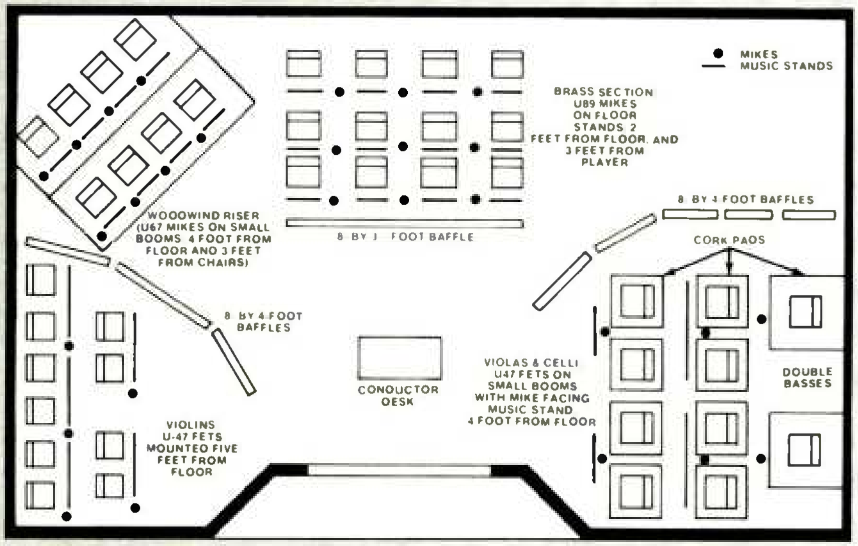
STANDARD MIKE LAYOUT AT "STAGE 10," UNIVERSAL STUDIOS



an overdub session would require a cue send so the players know whether or not they are in sync with the pre-recorded tracks.

Producers schedule all the music for television dates for one session, with drums, electric bass, guitars, etc. sharing the studio with more traditional acoustic instruments. In these instances, gobos between each section may be essential to preserve some semblance of isolation for later dubbing.

"I prefer to hear and blend with the other players as we're recording, and then depend on the conductor for direction." Sharp says. "But isolating the sections for TV dates makes it hard to hear the other musicians. Then we need the cue system. Speaking for myself, I like the headphones with just one side, so I can hear the rest of the orchestra with one ear, and the people in the section with the open ear. Intonation is so important, and that's one way to ensure that I can adjust."

Miking Suggestions

Miking suggestions from the three engineers interviewed for this article have been abstracted into separate, accompanying sidebars for easy reference. Each stressed that these are simply examples, and by no means constitute absolute solutions. The recommended placements simply happened to work at that specific date, or they usually work as starting points for most

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standard applications. The watchword here: *Experiment.*

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To many recording engineers, the considerations discussed in this article may appear unnecessary, or simply a hassle. Certainly with the current trend toward digital drum machines and synthesizers, working one-on-one with a player who is playing a direct-inject keyboard next to the console, seems the easiest way to go. But keep in mind that trends change rapidly. And only those who are well-rounded and prepared to adapt to fickle public preference will inevitably survive in the audio profession.

Of course, the argument arises that relatively few studios have the facilities to host a large orchestra, or even have access to that many musicians, either because of budget restraints, or simply lack of competent players. But the opinions, concepts, and techniques offered herein may be applied to live location dates, which occur regularly in most average-sized cities. In fact, with the right pre-planning, the industrious producer/engineer can schedule an overdub session on-location after a normal performance, and add those parts to the pre-recorded tracks back in the studio.

String dates may never become an everyday occurrence, but neither do they need become the rare exception. At least I hope not. ■■■

TELEVISION AND FILM DATES — *continued . . .*

Microphone Selection and Placement

Crofford also has the luxury of leaving the microphones pretty much in one place. "The mike choices and positions remain about the same. [See diagram below for mike selections and room layout.] Usually, when we have violas, they'll sit right next to the celli, and the basses go in the back. We've got cork pads under each of those players — not so much to isolate them from the floor, but to protect the floor against the stands that come out of the bottom of the instruments. The woodwinds are located toward the back left of the room, while the brass are closer to the center, with the horns in the last row, trombones in front of them, and trumpets in the first row."

TV Sound Techniques versus Feature Film

Although the positions and mike choices remain about the same, the exact distance from the instruments vary depending on the eventual medium for which the session is being recorded. "We don't mike close for a feature, unless it's a specific sound that the producer wants," Crofford says. "The sections are usually larger — 24 to 28 violins, 10 violas, 10 celli, and six basses. We'll retain a full complement of mikes — the same number for a closer-miked TV date — but raise them all up into the air to about 10 feet from the floor.

"Theaters play soundtracks much louder, so we need a 'bigger' sound. TV, on the other hand, is a tiny 'speaker, which requires that we treat everything pretty close. The closer you get to the instruments, the *brighter* the sound gets. By pulling away from the players, the instruments blend more, and create a fuller, 'lusher' sound."

Track Assignments

"In most cases we go direct to 35mm Mag, because we're always in a hurry," he continues. "The tracks are usually dubbed [remixed, and combined with other sound elements] the next day, and the show is on that night, or the next day. We have to do the final mix the first time through. Features, on the other hand, give us the opportunity to run 24-track and Mag at the same time, just in case the soundtrack is going to be released as an album. Then we remix at a later date.

"We have space on the board for 48 inputs, but we run out of space all the time. We generally lump all the violins on one channel, all the celli on another, and so forth. We try to get as many sections on their own tracks as possible, but we don't always have that much space." ■■■

WORKING WITH THE CORRECT ENVIRONMENT
A Conversation with Armin Steiner

At the age of 15, Armin Steiner decided that a career in audio engineering was for him, and he secured a job at Los Angeles' Electrovox Recording Studio, located across from the Paramount Studio lot. When he felt that he'd secured sufficient experience, Steiner went freelance and, ironically, has worked primarily in studios that he had built, including (in chronological order) Steiner Recording Studio, Sound Recorders (1965 to 1971), and Sound Labs (1971 thru 1981). Over that period of time Steiner has recorded, on a regular basis, some 135 artists, numbering among them 10 or 11 albums for Neil Diamond, all the Bread, Richard Harris, and Helen Reddy sessions, and various other artists such as Barbra Streisand, Dolly Parton ("Here You Come Again"), Jim Webb, Kenny Loggins, and many more. Currently employed by 20th Century Fox, Steiner handles soundtrack recording for the studio's television shows, such as *The Fall Guy, Trapper John, Nine to Five, MASH, After MASH, Manimals*, and *Navy*.

"When preparing for any string session," Steiner cautions, "the engineer should first pick a super studio with great ambient acoustics. To achieve the sound of a more classical nature, you need the depth, sonority, and the wonderful timbre of the instruments.

"There would be no need to use any kind of baffles when doing a session with just strings in the room. Actually, I would use screens only to record an orchestra in a very difficult acoustical environment, where the drums and brass are pounding, for example. The screens would go around the drums and guitars to separate them from the fiddles, as well as a couple more between the brass and violins. That would allow the string mikes to be raised a little bit higher, without being penetrated by the brass or rhythm instruments.

"The main objective is balance. If you're doing brass and strings at the same time, and the brass are going to overblow the strings, I don't care what type of acoustical environment you're in, you're *dead*."

Microphone Techniques and Selection

"Assuming we have melodic writing," he continues, "good harmonic background, and the proper acoustic environment, with a serious arranger who knows the voicings and how to balance an orchestra, recording of this category could be done with two microphones. I suggest using as simple a microphone technique as possible, so [that] the arranger or composer's music will come through and blend as he intended. But to mike this way, you *must* have the right mikes, and a room with maybe a one- to two-second decay [the studio at Fox has close to a two-second decay time], good distribution of high- and low-frequency information, and probably the higher the ceiling the better. Thirty feet or a little less is nice, but I've certainly done strings in rooms with 20-foot ceilings, or dead rooms with [added] reverb.

"That's where the engineer needs to know music, and what strings *truly* sound like. I've hardly ever seen engineers go out into the room and listen to the instruments or sections to learn the original sound. So many engineers base their work only on what they think they know through a loudspeaker, or a recording that someone else has done. They never even go to concerts to hear what natural instruments sound like. It's the universal ill that I have experienced with *very* few exceptions. [The interested reader is referred to Jimmy Stewarts two-part article, "An Engineer's Guide to Music," published in the June and August 1983 issues — *Ed.*]

"My complement of microphones is what's available to me at Fox: for strings, the original AKG 414s with C-12A capsules [in contrast to newer models, which use a different capsule]. Those are good on woodwinds, too. A wonderful group of mikes for brass are Telefunken U47s with Church electronics. Over the percussion I use Sony C-37s.

"I recommend picking up two or three omnidirectional mikes. I've had good success with the original AKG C-12As, Neumann M50s, and a variety of Schoeps omnis. Generally, I use two omnis, but sometimes we have the luxury to go with three channels and I'll set up the orchestra in classical manner.

"To use omnidirectional microphones, you must have the acoustical environment to take advantage of it, which means catching the back waves. Actually, an omni has more linear qualities than other types. It's a lot smoother, and not as strident.

"If I'm doing tracks in stereo, I set up two omnidirectional mikes in a typical stereo X-Y pattern, separated slightly to either side, and angled down about 45 degrees so that each pattern covers the strings equally, and picks up in sort of a hemispherical manner. Of course, the two patterns overlap, but you don't want half the music on one side, and the rest on the other. Our ears don't hear that way.

"Strictly speaking, most pop records made today are not really *true* stereophonic recordings — they're actually two mono tracks married together with delay and echo to *sound* like stereo. The definition of stereophonic [sound] takes into account certain phase relationships, time delays, and such that produce true three-dimensional stereo recordings.

"The omnis pick up the direct and reflected waves, which all combine so [that] the ear hears them naturally, as opposed to tracks that are directional. Depending on the size of the room, I would back those mikes up so they would focus on the overall section as a whole. The microphones may be about the same height, but not necessarily, depending on the environment: how much ceiling height is available; how far back the mikes are from the orchestra; etc. There are no given rules.

"A lot of the Jim Webb recordings [all Richard Harris' albums; for example, 'MacArthur Park'] were done at Sound Recorders with a single stereo microphone [Neumann SM-2, or SM-69] positioned 10 or 12 feet over the conductor's head, and slightly behind him. Occasionally, I'd use a third mike, which may sometimes add more mono than stereo if it's picking up too much information. That's fine for motion pictures, because you want only to spread out the sound

Room layout and string miking at Twentieth Century Fox Stage 1.

WORKING WITH THE CORRECT ENVIRONMENT — continued . . .

without too much directivity. It all comes down to having the *right* room. If the room is dead, you'll get no reflections or time delays —essentially the depth —on tape."

Adding More Mikes

"If you're dealing with an arranger who writes a lot of notes that requires a good degree of articulation for the inner voices, yet the environment doesn't support that clarity," Steiner offers, "you may have to go to more of a multimike technique — maybe three or four mikes over the violins; one very high over the violas; and a couple over the cellos [all cardioid-patterned AKG models].

"When using more microphones, they wouldn't necessarily be as high up as the first case we discussed, although they could be. If they are, the music should be a higher order of rhythmic definition. But if you're dealing with a very busy piece of music, then the mikes have to come in a little bit to hear all that 'busyness.' Distant miking won't preserve the order of definition, because the other elements in the rhythm section are sort of pounding away at the strings. Let me emphasize that the *music* must always be the guiding force, and dictates what approach to use. That's the bottom line.

"Sometimes, smaller string sections need more reverb, and to compensate we have to assign two or three players to a microphone. Those mikes can be anywhere from six to eight feet above each one on two stands, and facing down at a slight angle of about 30 degrees [usually two players per stand]. The celli, violas, and basses are miked the same way. You're really going for the group of two or four players: two in front, and two behind."

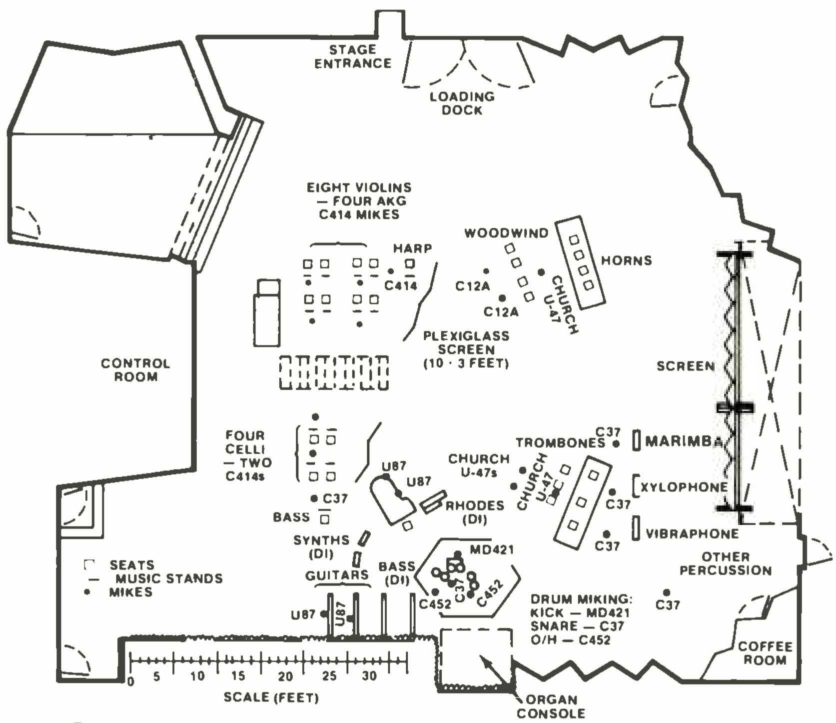
Television Dates

"Keep in mind that television recording is done in one pass with sometimes one rehearsal," warns Steiner. "In several cases we never even rehearse. If the cues are similar, we do as much as a half hour of music in four to four and a half hours [of recording time].

"Live recordings for TV seldom afford the luxury of a balanced orchestra. We usually end up with three trumpets, three trombones, four french horns, six woodwinds, a pounding rhythm section, eight violins, and four celli — if we're lucky! In many cases, we're not necessarily going for a 'pretty sound.' Under those circumstances, I mike each stand of violins. If we recorded the date in a classical set-up, the drums and brass would have to play very lightly in the room, because we really don't have the 25 violins needed to compensate or balance.

"The celli are often playing rapid passages [for action scenes] that are being doubled by the trombones and the french horns. For the producer to hear the celli, the mikes must come in close, or else all you'll hear are the brass."

BASIC ROOM LAYOUT AT TWENTIETH CENTURY-FOX STAGE 1
ENGINEER: ARMIN STEINER



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WORKING WITH THE CORRECT ENVIRONMENT

— continued . . .

Track Assignments

"Production for a weekly television series happens so fast that we knock out the entire project in a day," he says. "Sometimes we can take a little longer for a mini-series, or something that doesn't go on the air the day after scoring. But, in either case, the music is usually left until the last moment. Dialog is supreme, and sound effects are important . . . music is last. If a string passage gets in the way of dialog the music has to be brought down.

"For normal television dubbing purposes [on to three-track 35mm Mag stock], the center channel — or channel two — is devoted to the rhythm section; channel three is strictly woodwinds and brass; channel one is any and all strings. Obviously, we don't get 100% isolation, and sometimes a little brass gets in with the strings. But that's minor.

"The exception to that assignment is *Dynasty*, [for which] the instrumentation is mainly strings and woodwinds, with maybe a little piano. I split the strings left and right: violins on channel one, violas and cellos on channel three. The woodwinds, piano and tympani go on the center channel.

"Now a stereo picture, or one that is fully orchestrated for 50 or 60 musicians, gets set up in a clasical layout. Assuming there is no rhythm section, I approach it as though it were a legitimate date. I make a left, center, right stereo mix, so the dubbers can control the high and the low ends independently. Let's say violas, cellos, and low brass on channel three; upper channel [#1] would be fiddles, and trumpets; and in the center I'd have woodwinds, percussion, and french horns. That's pretty much what the actual orchestra would sound like live.

"I get calls from the outside all the time just to sweeten records with strings and brass. If somebody brings in a multitrack tape with just three open channels, I'm helpless! yet I generally use only two channels for strings anyway, unless I split it out on three tracks. I go for a stereo perspective with high and low strings just as I would for a stereo picture [violins on the left], regardless of whether or not the miking is a little bit closer than normal."

Outboard Effects

"Reverb is the engineer's best tool to create the depth that the string section should have," Steiner offers, "especially in deader rooms. In the old days, when we were dealing with only four fiddles, I would use tape reverb. A lot of engineers prefer to put that delay in front of the chamber. What they are doing is trying to duplicate the first reflection in a good room. Now they're using digital delay lines, so you can actually delay the first reflection, which defines the 'character' of the reverb.

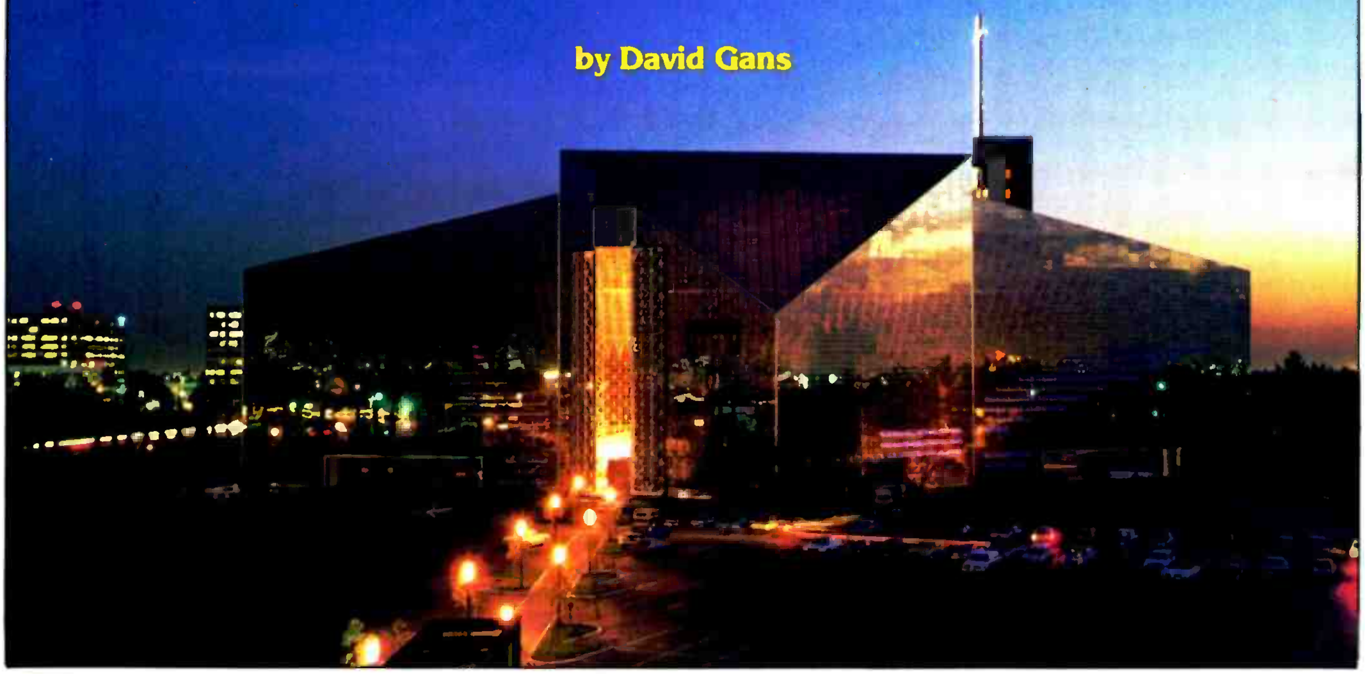
"Personally, I believe in using a relatively long delay, and not using very much of it in the mix. Most guys use a very short decay time, especially when they're running through an EMT, where you can vary the amount of decay, and drive the unit hard. I've never believed in that. I am a proponent of using at least a reasonable 3- to 3½-second decay time, and then feeding in as much as I need to sound musical, and still give me a big room sound. The music will dictate that; the music must *always* be the guide.

"I set the tape delay at 15 or 30 IPS [approximately 133 or 66 milliseconds, respectively, if the tape heads are two inches apart]. I've even set it at 7½ IPS [266 milliseconds] with just a teeny amount of reverb for some of the Bread records. That's quite a drastic, wide echo, and has to be used sparingly. But the music allowed me to do it. Fifteen IPS is much easier, because the delay isn't so drastic. Sometimes we had to do that when the chamber didn't have a long enough delay time. Sound Labs had a chamber with about a 4½-second decay time. It was wonderful on strings and brass, because it recreated that 'floating' quality, so records didn't sound like demos. Today's music is *much* to dry." ■■■

CRYSTAL CATHEDRAL

Overcoming Reverberant Acoustics to Provide Sound Reinforcement for a Problematic Environment

by David Gans



"Third time's the charm" . . . "Less is more" — the clichés come trippingly to the tongue in describing how McCune Audio/Visual and Meyer Sound Labs help solve some of the sound reinforcement problems of Southern California's Crystal Cathedral. But the solution itself is anything but a cliché; it uses the brand-new technology developed by Meyer Sound to increase intelligibility with a few loudspeakers, where previous design philosophies had bludgeoned the acoustical environment with massive arrays of speakers, and banks of delay lines.

From outside, the Crystal Cathedral is an impressive sight, its huge angled planes of mirrored glass forming a striking addition to the generally low-slung skyline of Garden Grove, south of Los Angeles — although some people might think it more appropriate to the architectural scheme of Disneyland, a few miles deeper into Orange County. The complex supporting framework makes the Cathedral breathtaking from the inside, too, but the combination of glass and metal results in an acoustical nightmare of reflected sound. It took three tries, and more than $2 million, to solve a unique set of sound reinforcement challenges.

"The architect didn't want to see any speakers hanging from the ceiling or on the walls," explains Randy Mobley of the Crystal Cathedral staff. The sound system that had been installed when the Cathedral opened in September of 1980 was a pew-back design, with many small speakers mounted a few feet apart throughout the main floor and the three large balconies.

"We heard they were having severe sound problems," recalls Ken DeLoria, of San Francisco-based McCune Audio/Visual, which has a branch office in Anaheim, not far from the Cathedral. "We made an appointment, and one fine afternoon we went and took a look at what they had."

What the McCune crew saw was "a really overbuilt system, with too many speakers, and too much fancy switching," DeLoria explains. "It was designed around imaging; they wanted to be able to change the apparent location of the sound source," from the lectern to the choir loft, etc., at appropriate times. "It was a mess; there were something like 60 delay lines to time-correct the pew-back speakers, and far too many sound sources for such a reverberant environment."

With the sound coming from so many different locations, DeLoria adds, "in any given seat one person is hearing at least five or six sources — maybe as many as 10 — and they're all different distances away, so they're at different time intervals. It's like having five or six DDLs, each set at from 10 to 40 milliseconds, ganged up. It's not going to enhance intelligibility in a difficult environment — it's going to *destroy* any chance for it."

The pew-back system was replaced fairly early in the game, according to Mobley, with a custom system consisting of three main "Crystal Clusters" —plexiglass cabinets loaded with Altec drivers, one cluster per balcony — and more than 20 sets of bass cabinets and horns distributed across the ceiling. "On paper, it's not too bad," DeLoria concedes, "if you ignore the fact that sound doesn't stop when it hits the floor.

"It's not like a nice little lighting plot where you overlap all your dispersion patterns, or have them just meet each other. When sound hits the floor and bounces back up again, or hits the walls, you're just compounding the problem. You end up with random sources coming from many different places, reflecting and continuing to reflect."

The Crystal Cluster system suffered from intelligibility problems, just as its predecessor did. "When people are trying to listen and they can't understand, their attention spans eventually break down," Deloria continues. "When they only catch every other word, they're missing the whole point of being there, which is to receive some sort of communication. It's irritating, and finally people give up. At the Crystal Cathedral, they were routinely losing whole balconies; the people would get up and walk out in the middle of services."

Needless to say, Dr. Robert Schuller and his staff and congregation are relieved that this problem finally has been solved.

Coping with Reverberation

McCune's original Crystal Cathedral installation, assembled from the company's rental stock, consisted of six Meyer Sound Labs two-way UPA-1 loudspeakers. The 180° coverage of this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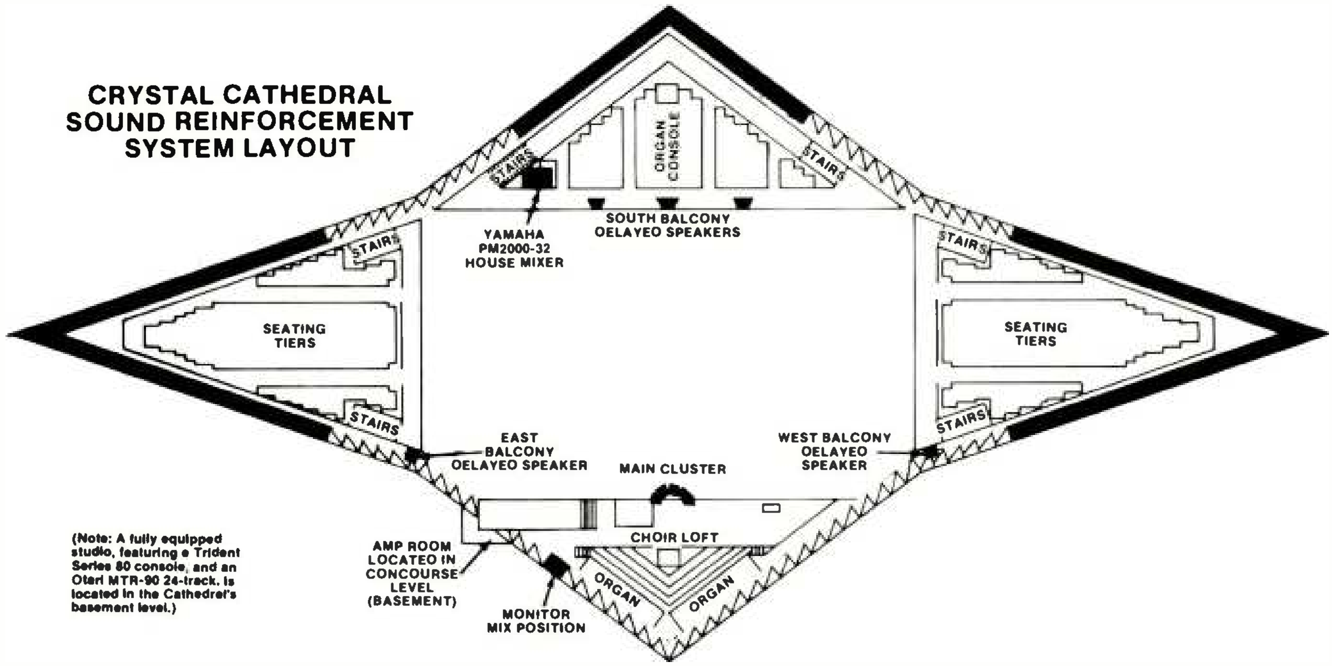
CRYSTAL CATHEDRAL SOUND REINFORCEMENT SYSTEM LAYOUT



array "put a little too much sound into the side glass panels where there weren't any seats," DeLoria recalls. "By reducing that to five cabinets, we got a corresponding reduction in the amount of reverberant field that we excited — and therefore improved intelligibility."

Another lesson learned in the first two weeks the rental system was up in the Cathedral was that the single cluster "left a little bit to be desired in the balconies," according to DeLoria. An additional set of three UPA-1s was hung from the ceiling at the front of the large south balcony, with a digital delay to bring their output into line with the sound from the mains — and a single delayed UPA was installed for each of the two smaller balconies. Each of the three speaker systems has its own Meyer processor. "These extra UPAs are operated at very low levels," says DeLoria. "Each has its own 27-band graphic, with most of the low-end rolled off. They just add a little bit of presence because, by the time you get that far away from the main cluster, there's so much reverberant field in the room that we found it necessary to add the local sound sources.

"The key is that they *augment* the main cluster, supporting the single source rather than becoming a whole new source operated at high levels."

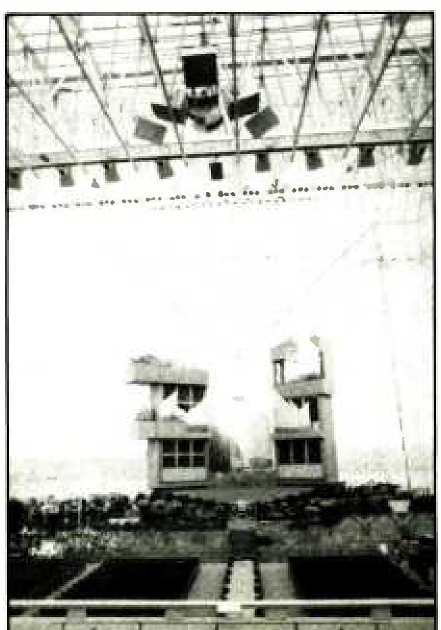
The Crystal Cathedral's reverberation problem wasn't the only challenge faced by the sound system designers; there's also the matter of Schuller's extraordinarily dynamic delivery. "He will stress a couple of words or syllables heavily, and then follow immediately with a word that's barely audible," DeLoria explains. "The result is that the words he [says] with so much level will excite the reverberant field in the room — and it'll still be responding to the loud phrase when he follows with the soft word.

"One of his favorite phrases is, 'Tough times don't last — tough people do,' which he delivers with a lot of sibilance and sharp consonants." DeLoria demonstrates, enunciating the phrase with percussive, sibilant "T" sounds, and a loud, rising falsetto on the first syllable of "people," then ending with a barely voiced "do."

"The last word would be completely masked by the reverberant field resulting from '*PEE*-ple,' " DeLoria explains. "It really compounds an already difficult problem." For this reason, DeLoria concedes that a certain amount of signal processing gear is essential — but the extraordinary clarity of the Meyer speaker system demands as unaffected signal as possible.

"There really are a lot of problems you can get yourself into when you have too much in the signal path," DeLoria amplifies. "Especially when you have one band being boosted on your board, and then the same band with maybe a narrower Q [bandwidth] being cut on the graphic, boosted again at the [Audio + Design] Vocal Stressor, and then cut at the de-esser, boosted at the [Aphex] Aural Exciter — then it goes down to the amp rack, and gets boosted and cut again to match with the loudspeakers."

Main Altec speaker cluster suspended above the choir and organ loft in distance, and Meyer delay speakers for south balcony (top).



Photography by David Gans

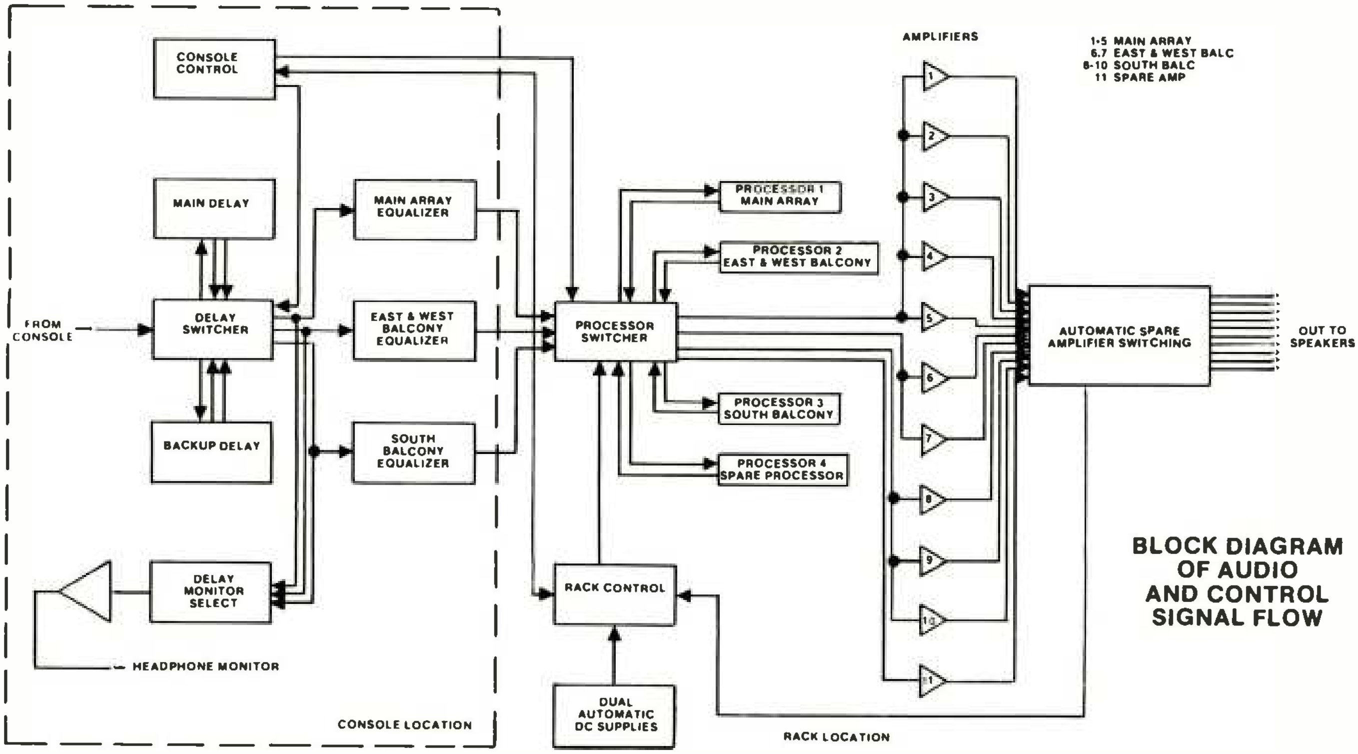
The question arises as to how anyone can believe that a reverberation problem can be dealt with effectively by using large amounts of hardware. "It defies logic," DeLoria sighs, "but it is a common belief." He concedes that it's hard for a manufacturer to refuse a contractor or user who specifies a lot of speakers when a few will do, because the manufacturer is in the business of selling his goods.

McCune has long based its business on doing on-site design work and testing. "I think we're going to see more of that in the future," DeLoria notes. "We're going to have to prove a sound system can perform before the customer commits to a large capital outlay.

"You could *never* test a pew-back system or a large distributed horn system — and that, in itself, tells you how silly those complex designs are, relying on miles of conduit and putting horns all over the place. There's *no* way to determine whether they're going to work or not before they're installed.

"Multiple sound sources don't overlap like multiple light sources do. They reflect and cause all these different echoes and time delays, phase aberrations, and so on."

"I don't know why it is, but rental systems always seem to work better than systems installed by designers and contractors," comments McCune's general manager, Mort Feld. "What we do is install a rental system, and if the cus-



BLOCK DIAGRAM OF AUDIO AND CONTROL SIGNAL FLOW

tomer likes it we say, 'Here's what it takes to duplicate it.' "

Feld and DeLoria are in complete agreement on the point that the Crystal Cathedral installation would be impossible were it not for Meyer Sound's speaker systems. "John Meyer builds his [units] with no compromises," Feld offers. "His distortion figures are unbelievable, and the response is amazing; his loudspeakers are the only ones that can do what they do." [The principle of single-point source, phase-coherent speaker arrays is explained in the article, "Time Alignment of Sound Reinforcement Equipment, by Patrick Maloney; December 1980 issue of *R-e/p — Ed.*]

By the time the McCune people tackled the Crystal Cathedral project, DeLoria says, "all the basic cabling, conduits and signal path engineering had already been done, and seemed to be working quite well. They already had a 32-channel Yamaha PM2000, and a lot of front-end gear, including parametrics, graphics, a Vocal Stressor, and so on. We provided only an amplifier/loudspeaker package with front-end processing, including some new equalizers and, of course, the Meyer processors."

"Signal processing sucks," says Feld bluntly. "It buys more damage than advantage. We like to keep it straightforward and simple." Nevertheless, unique properties of the Crystal Cathedral — and its principal orator's style — require a certain amount of compensation.

For example, the main speaker cluster has a parametric equalizer, which is considered very useful for removing the low-frequency buildup characteristic of highly reflective environments. "With the parametric you can dial out some of the presence peaks that the room seems to have, and add a little bit of high-end in the right place to compensate for what tends to be an excessive bass problem," DeLoria explains.

"A 'flat' speaker that responds very well in a normal environment tends to sound a little bass-heavy in the Cathedral. A single UPA all by itself seems to have too much bass; the first impression is that there's something wrong with the horn, that the high-end driver isn't working right," he continues. "The central cluster needs quite a bit of low-end rolloff in order to perform correctly."

Power Amplifier System

The amplifier rack assembled by McCune for permanent installation at the Crystal Cathedral includes several features designed to prolong the life of the system, and ensure a minimum of inconvenience when something does go wrong. Magnetic circuit breakers — one for each amplifier, Meyer processor, power supply and blower, and even one for the utility outlet on the front panel —respond much more quickly than the standard variety.

McCune Sound's Ken DeLoria with partially assembled permanent amplifier racks.

By running the main line voltage through large resistors (which actually are heating elements) when the system is powered up, the amplifiers are brought up to about half voltage for two seconds. "This 'step-start' feature prolongs the life of the power supplies, and it also keeps the magnetic breakers from tripping due to the high inrush of power when the system is turned on," DeLoria explains.

Status lights on the main control panel enable the operator to see whether the power distribution system is operating correctly. From here, the spare Meyer processor can be switched into the place of one of the three on-line processors; there's a 100-millisecond delay built in so that the power can stabilize in the processor before it is connected to the UPA speaker. Test points are available at the control panels for all amplifiers, speakers, and processors; there's also a test input common to all four processors, so they can be examined and adjusted simultaneously. A "Master Speaker" disable switch trips all of the speaker protection relays, enabling the amps to be operated off-line for service and testing.

A "Zone Disable" switchboard shuts down the speaker system for any or all of the balconies. "If the house isn't full, one or more of the balconies can be closed," DeLoria notes. "By turning off

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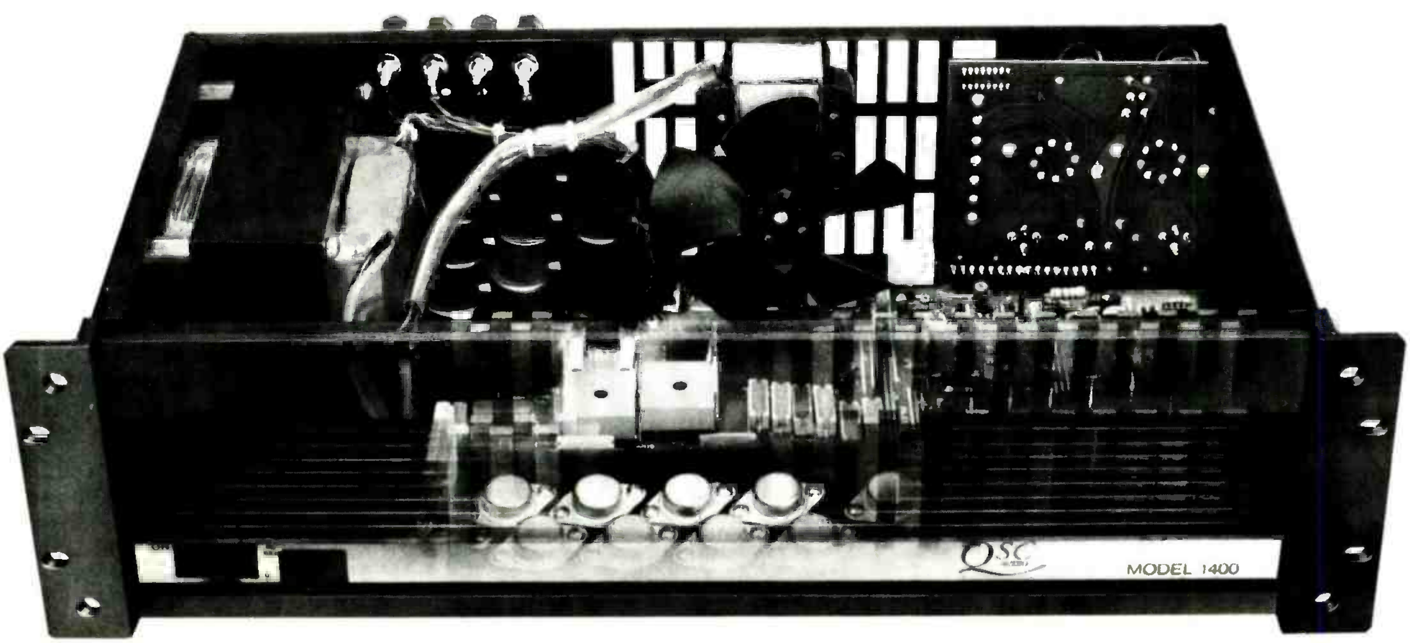

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the speakers up there, reverberation is reduced in the whole building. It's subtle, but it's effective."

There are two power supplies for the system. One is normally on-line and the other powered off; if the on-line supply fails, the backup is automatically switched in. A manual override allows the backup supply to be used on-line periodically, which McCune recommends be done several times during the first year in order to "burn in" the second system.

The automatic spare amp switching panel (a.k.a. "McCune Auto-Correct") switches a spare amp into any of the 10 positions (one for each UPA), should an on-line amp fail. Status lights glow on the control panel to inform the operator that the system has adjusted itself, and indicate which amplifier failed. Each of the custom McCune/Hafler solid-state power amps (one per speaker, each providing 320 watts RMS into 8 ohms, with FET output devices) has an identifying plate on its face with a schematic diagram of the Cathedral's speaker layout; the UPA to which each amp is normally connected is indicated by a darkened square. "The serviceman or operator doesn't even have to speak English to know which speaker is off," jokes DeLoria.

Other safety and convenience features include thermostatically controlled cooling fans. As temperatures increase, so does the speed of the fan; the life of each fan is prolonged because it doesn't run any faster than necessary.

A remote control panel at the mix position in the south balcony duplicates all monitoring and operating functions, except testing and Master Speaker Disable, which are only needed in the basement room where the rack is located.

"The system is designed to be well-behaved," says DeLoria in conclusion. "It lets you know what's going on, and even though no piece of equipment is pushed too hard and everything is protected, there's a spare ready to go on-line immediately, so there's virtually no downtime."

Keeping it Simple

John Meyer points out that properly interfaced, the failure rate of MSLI's gear is approximately 0.5% for electronics, and 1% for speakers — and that figures such as these make it hard to justify an investment in complicated monitoring circuitry. "You're much better off with two complete systems than with one system, and a [device] to tell you it isn't working," he says. "And it's hard to make a monitoring system smart enough to distinguish between catastrophic failures, and trivial distortions such as momentary clipping. An inadequately designed protection circuit could shut down the entire operation because of a momentary glitch."

It is Meyer's opinion that "The extra

House mix position, with Yamaha PM2000 console, and signal processing equipment.

features added by McCune [in the Crystal Cathedral installation] are not necessary to the operation of the system. We feel that the simpler a system is, the more reliable it is likely to be." Meyer prefers to create straightforward systems with amp-and-speaker channels "interlaced" in the array, so that if one set fails the net result is a manageable decrease in overall performance, rather than total silence in one section of the field of coverage. "By making the system simple and internally redundant, we can see to it that a failure won't knock the whole thing out, and stop the show."

Meyer expresses some irritation at the "mystification" of his products by contractors who use them. "We've always fought the 'black box' syndrome," he says, "and when they mystify their additions, they mystify *our* systems as well. Our position has always been to be very clear and forthcoming about what we are doing.

"Proprietary, one-off devices that are not manufactured and commonly available are troublesome, because they can't simply be replaced when they break down." Such devices are almost never properly documented, Meyer notes, "so if the designer isn't available then the user is stuck." By contrast, when an element in the Meyer system breaks down, "replacements are readily available and fully documented."

Recognizing the value of field testing and demonstration is essential to the continued health of the sound-reinforcement trade, says Meyer. That's one of the reasons why Meyer Sound Lab has developed systems with the rental market in mind, rather than concentrating on sales of permanent installations. "We know that sales will follow *demonstrated* results, and we can learn something from every installation in which our gear is used," he says.

Differences over the specifics of hardware aside, Meyer Sound and McCune Audio/Visual agree that all the plans, diagrams, and specifications in the world do not mean as much as the on-line demonstration of an efficient sound system, in place at the intended site. The absolute bottom line, be it a race track, a rock concert, or the Crystal Cathedral, is a reliable, efficient system that *sounds good.* ■■■

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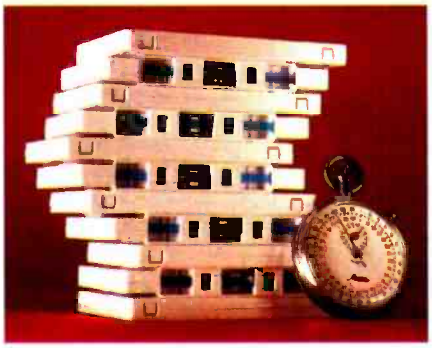

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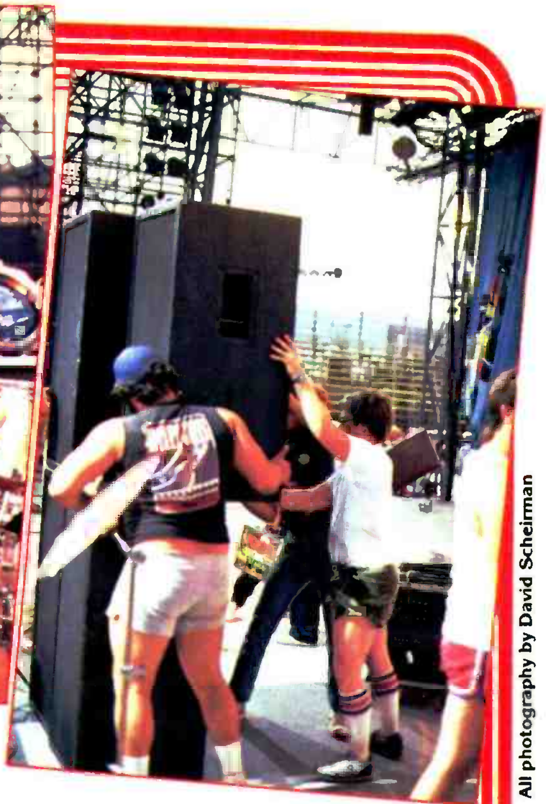

All photography by David Scheirman

THE 1983 US FESTIVAL

COOPERATIVE SOUND SYSTEM DESIGN AND OPERATION BY CLAIR BROTHERS AND SHOWCO

by David Scheirman

The 1983 US Festival, held last May at Glen Helen Regional Park in San Bernardino County, California, may not have drawn the largest outdoor festival crowd ever assembled, but those persons in attendance did witness one of the planet's most massive outdoor concert sound systems ever pieced together. Staged over the Memorial Day weekend in a 54-acre bowl scraped out of a hillside in the Los Angeles-area public facility, the US Festival had an $18 million budget, which contained ample resources for the best in sound, lighting and staging. The Festival was produced by the UNUSON Corporation (An acronym for "Unite Us In Song"), which in turn had been set up by Steve Wozniak, the multimillionaire inventor of the Apple personal computer.

Wozniak explains his criteria for choosing a Clair/Showco combination to handle sound for the Festival: "*No* compromise . . . period! That was it. We shopped around until we found someone who could give us exactly what we wanted without trying to change things, and who was willing to work with us. And that's what we got. This is essentially the same system which we used for last year's 'US 82' Festival, but I understand that there have been some technical improvements."

An outdoor site such as the one prepared for the US Festival presents a sound system designer with some very real problems to solve. Outdoor sound requirements for crowds numbering up to 300,000 people are very different from a typical arena tour. One problem in an arena so vast as the US Festival's 54-acre site is the speed of sound. Since sound waves do not travel quickly enough to provide simultaneous delivery of the music at both the front and rear of the audience area, a series of speaker towers that receive a delayed signal is necessary.

Temperature gradients present a problem of a different sort; since the speaker scaffolding is stacked up into the air, sound waves encounter layers of warm and cool air. The various alternating (and constantly shifting) layers in the atmosphere cause changes in the system's frequency response. Typically, those loudspeaker units at the top of the stacks will require a more pronounced emphasizing of low and high frequencies. All of which presents the need for a far greater amount of control over the system's equalization than would normally be the case in a smaller, indoor system.

Clair House System

Main house speaker stacks for the US Festival were provided by Clair Brothers Audio, of Lititz, PA. Situated on sound wings which were 184 feet wide — the entire stage platform from side to side totalled 436 feet — the Clair house system comprised 180 S-4 speaker cabinets stacked 10 wide and three high, in three different tiers on each side of the stage (Figure 1). Additionally, a fourth upper tier of scaffolding was added that contained 32 long-throw high-frequency horns per side.

Roy Clair explains how the US Festival system was assembled, and details some of his philosophy for outdoor speaker systems: "This year, we were quite lucky. A lot of our tours ended, which gave us a lot of available gear during this time frame. I can't say that we planned it this way, because you certainly can't go around telling your accounts when to go on the road, but it did work out well.

"We already had several truckloads of gear here on the West Coast, which

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logistically worked out for us. Last year, we scheduled [the Festival] as best we could, and we got help from other people. This year, we did not have to farm out any parts of the system, although we do have Showco providing the delay towers again. Last year, Clair had quite an advantage during the bidding, politically speaking, since a majority of the acts on the bill were our touring accounts already. So these acts requested that Wozniak use Clair. We are here again this year primarily because everything worked out so well last time."

According to Clair, the US Festival tied up sound equipment for more than three weeks, although the Festival involved only four days of actual live music. "The time involved for transportation, the week of setup time and system testing, plus the four-day break between the rock and the country performance days, really stretched it out," he says. "We have many members of our staff who have been working on just this project for more than a month. Our advance-notice time was about three months; then, coordinating the on-site arrival of all of the gear, and assembling four or five different road systems into a single large cohesive system, was the big job."

Design of the system was done by Clair Bros. after UNUSON informed the company of the basic parameters: crowd size; budget; and site dimensions. "Our cabinets are such that the more of them you stack, the better they get," Clair offers. "But, with horn-loaded

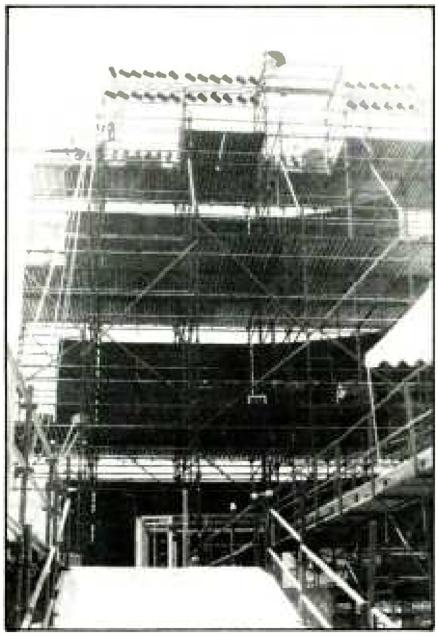


Figure 1: Clair house system comprised 180 S-4 cabinets and 32 long-throw constant-coverage long-throw horns; S-4s were stacked 10 wide and three high on three tiers per side.

cabinets, that is not always the case. What we have done here is to create massive line arrays of the same type of component, so that the '18s all line up —the horns, and so on. The composite boxes are like building blocks; they make it very easy to assemble a large system."

S-4 Speaker Cabinets

Since 1971, Clair has been building and using the composite loudspeaker cabinet known as the S-4. This cabinet was considered by many to be unique for its time, and still is the standard by which other composite units are judged. More than just a new packaging concept, the S-4 was engineered to break the concert sound system down into "modular" units; one speaker cabinet contained all of the components that would be found in the entire sound system.

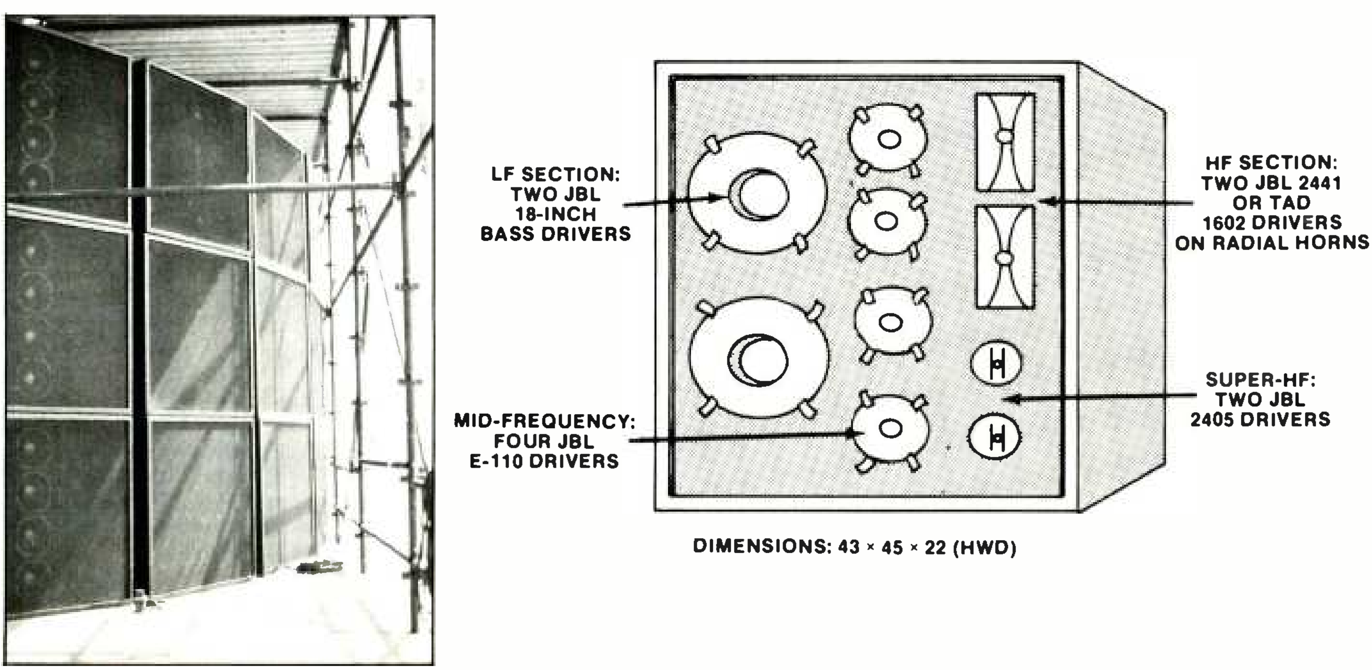
The S-4 is a four-way enclosure, containing two 18-inch cone drivers in the low-frequency section; four cone drivers in the mid-bass region; two compression drivers mounted on horns in the midrange section; and compression-type HF units on the top end. Type of speakers used varies from cabinet to cabinet, and five or six different variations of the S-4 have been produced in the past 12 years. However, the basic design has remained constant.

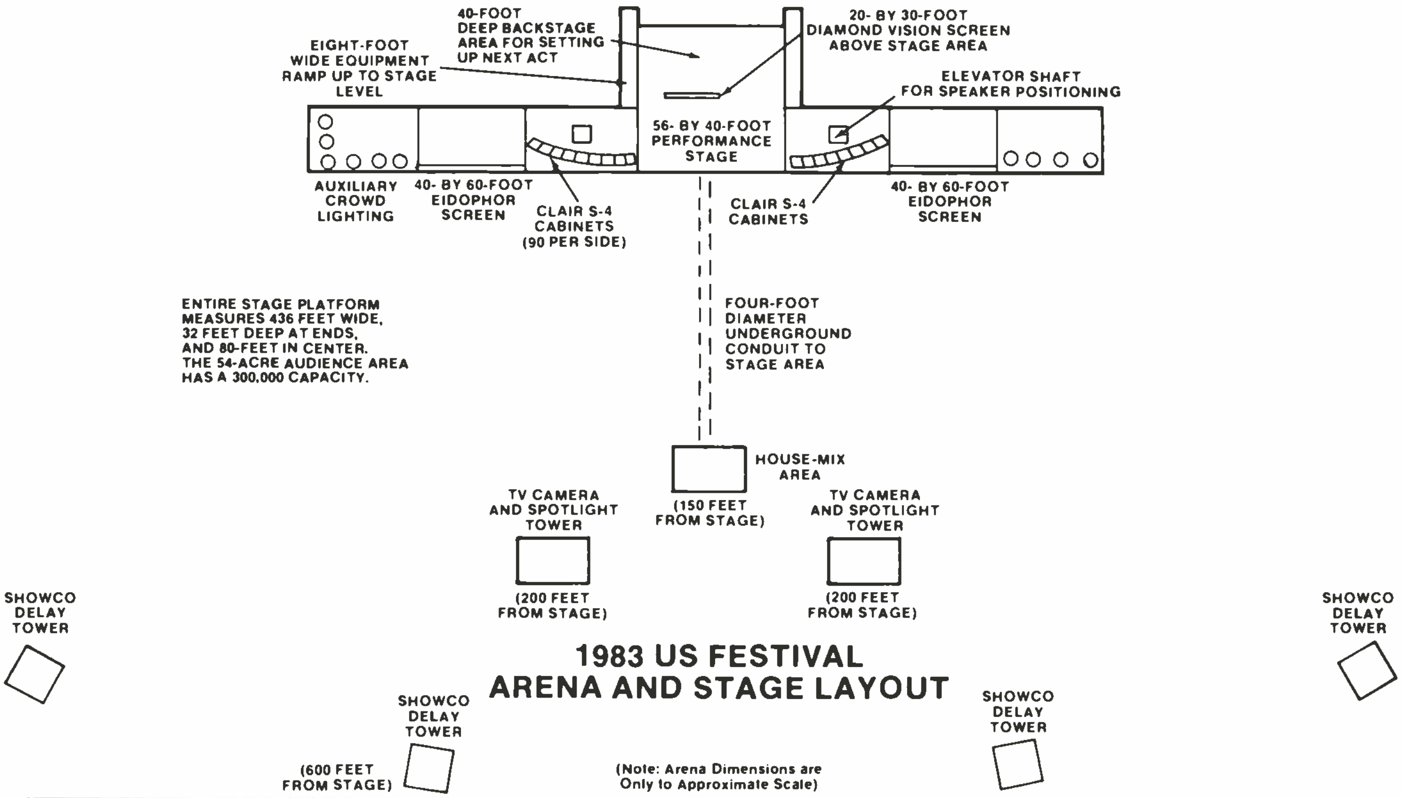
"We were really pioneering a new concept when we first assembled the S-4s," Clair concedes. "We wanted something that would pack well in the trucks, that could be easily handled, that looked clean, and would give all of our systems a consistent sound. We are constantly trying new transducers, new horn designs, but I think the all-in-one box has proven itself to be the most efficient way to carry a concert system from city to city."

The S-4 weighs approximately 425 pounds fully loaded, and measures 43 by 45 by 22 inches deep. Each type of speaker component within the cabinet is grouped in such a way as to create line arrays when the cabinets are stacked (Figure 2).

Each of the three tiers of S-4 cabinets on each sound wing was split into two, separately-controlled systems: half were pointed straight ahead; and the remaining half on each level were given a gentle hemispherical curve to the outside, such that the outermost cabinets in each stack were angled away from the

Figure 2: Transducer alignment of Clair S-4 cabinet, shown left on US Festival stage, and right to illustrate driver configuration.





1983 US FESTIVAL ARENA AND STAGE LAYOUT

inside cabinets at approximately 45 degrees (Figure 3). According to Clair, each half of each tier had separate crossover feeds, and the input signal to each crosssover was processed through the company's own third-octave equalizer. With each side having four tiers, this gave the Clair engineers eight different sets of crossover and equalization controls on each side. (The left stack, for example, was split into eight banks of speaker cabinets, with one output mix going to eight sets of system output electronics.)

"We found it to be really important that the engineers have complete control over what was coming out of such a large loudspeaker array," relates Clair. "With this set-up, the lowest tier can have its level attenuated greatly so as to not harm the ears of the ground-level crowd. The middle tiers can be adjusted as needed to provide more or less of a given frequency range. And the top tier can be set to throw the sound all the way back to the rear of the site."

System specifications for the US Festival called for 90 to 100 dB average sound pressure level of program material at the rear concession-stand area —a distance of nearly a quarter-mile.

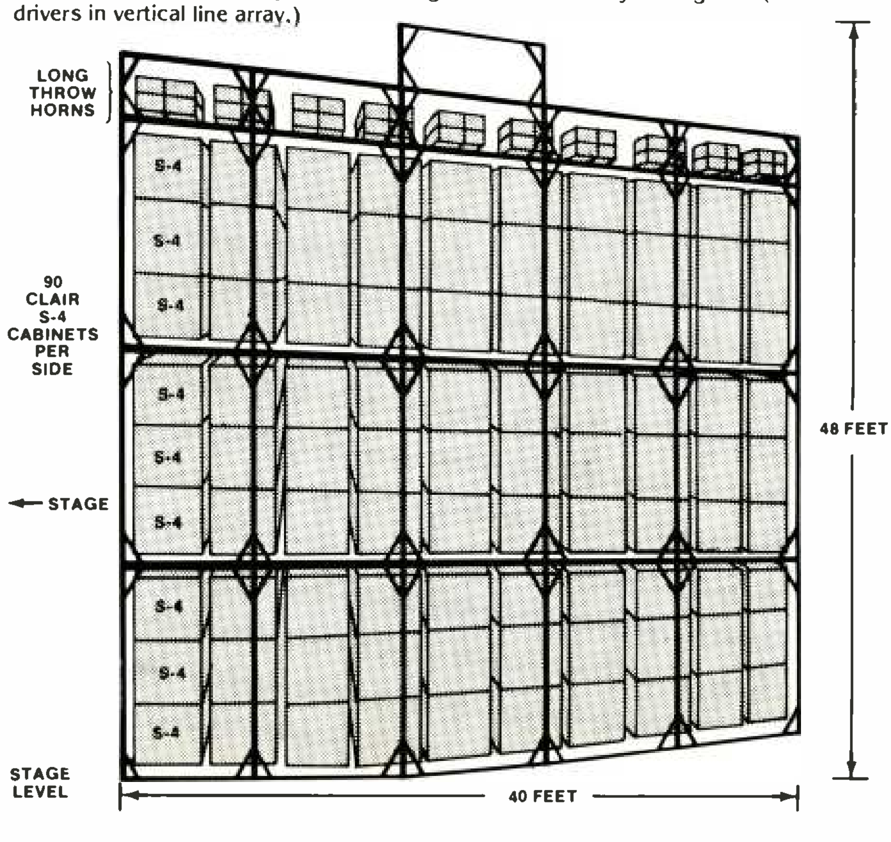
Long-Throw Horns

The fourth tier on each speaker stack was used for the long-throw horn array, the 20- by 40-degree fiberglass horns being packed four to a box. With a throat length of 18 inches, and a mouth size of approximately 6 by 8 inches, the horn —manufactured for Clair by Community Light & Sound — represents a new concept in long-throw arrays.

"This is one thing we have added to the system since last year's US Festival," Clair remarks. "This type of design is available from several different manufacturers . . . JBL calls their horn a Bi-Radial; Altec calls their's a Manta-Ray. I believe Electro-Voice first used the term 'constant-coverage'... it's a tradename. After last year, we found that the high frequencies were the most difficult to get out there, so we are using over 60 of these new horns made to our specifications. And [the result] is like night and day . . . it *really* helps.

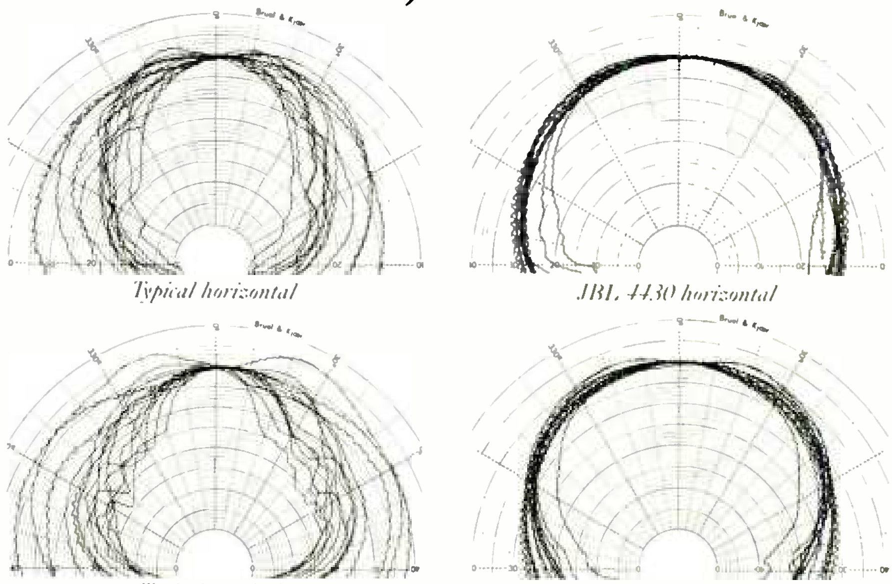
"I must say, that, for the whole industry, constant-coverage horns may not be

Figure 3: Speaker cabinets were laid out in a gentle curve so that the outside of each speaker stack angled away from the straight-ahead stacks by 45 degrees. (Note cone drivers in vertical line array.)



Before you invest in new studio monitors, consider all the angles.

Polar response comparison of a typical two-way coaxial studio monitor and JBL's new 4430 Bi-Radial studio monitor from 1 kHz to 16 kHz.



Typical horizontal *JBL 4430 horizontal*

Typical vertical *JBL 4430 vertical*

No one has to tell you how important flat frequency response is in a studio monitor. But if you judge a monitor's performance by its on-axis response curve, you're only getting part of the story.

Most conventional monitors tend to narrow their dispersion as frequency increases. So while their on-axis response may be flat, their off-axis response can roll off dramatically, literally locking you into the on-axis "sweet spot." Even worse, drastic changes in the horn's directivity contribute significantly to horn colorations.

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1. Patent applied for.

And the Bi-Radial horn's performance advantages aren't limited to just beamwidth control. The horn's rapid flare rate, for instance, dramatically reduces second harmonic distortion and its shallow depth allows for optimal acoustic alignment of the drivers. This alignment lets the monitors fall well below the Blauert and Laws criteria for minimum audible time delay discrepancies.

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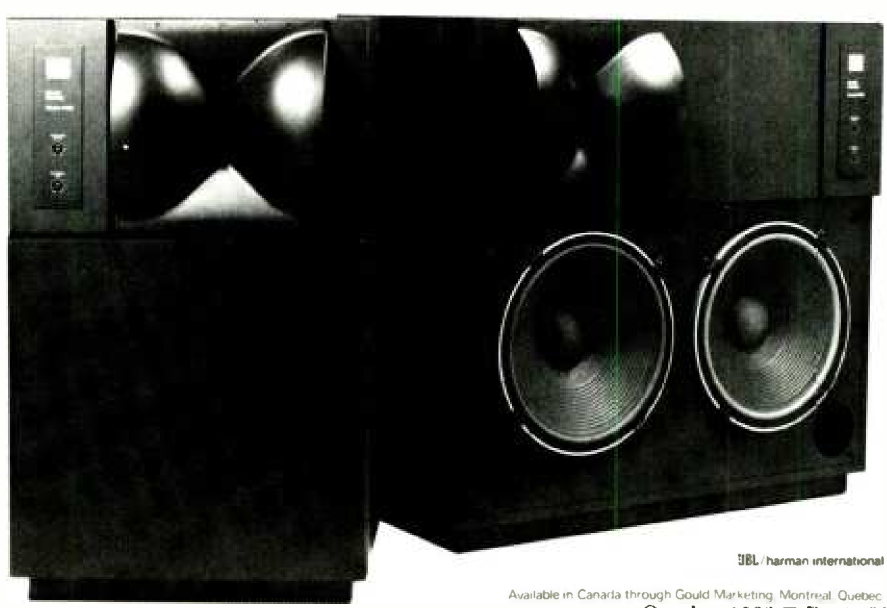

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the answer, but they are certainly an improvement over that typical 'honk' which you get from a traditional radial horn."

Power Amplifier Racks

Clair's S-4 system requires one amp rack for each four speaker cabinets. Each rack contains four stereo power units: Phase Linear 700-Bs, or SAE 2600s, depending upon how recently the rack was assembled. "The 2600 amplifier develops about 600 watts per channel into 4-ohm load," advises Clair engineer Mike Wolpert. "We have modified each individual amp in our shop. Each unit you see that has the Clair faceplate has been altered and set up for our particular use. The channel which runs high-frequency components puts out less power, about 400 watts ... it has special protection circuitry for the high-end devices to safeguard them from being destroyed due to excessive voltage."

The amp racks were placed on the sound wings in groups of four. Intake and exhaust fans on each rack were placed under a plywood enclosure that served as a cooling chamber. Large chunks of dry ice were set on a two-inch thick pad of styrofoam, and the chilled air contained within the plywood box allowed to recirculate through the amplifier racks, thus ensuring cooler operating temperatures (Figure 4).

With more than 50 amp racks in use for the main speaker stacks, thousands of pounds of dry ice were required daily.

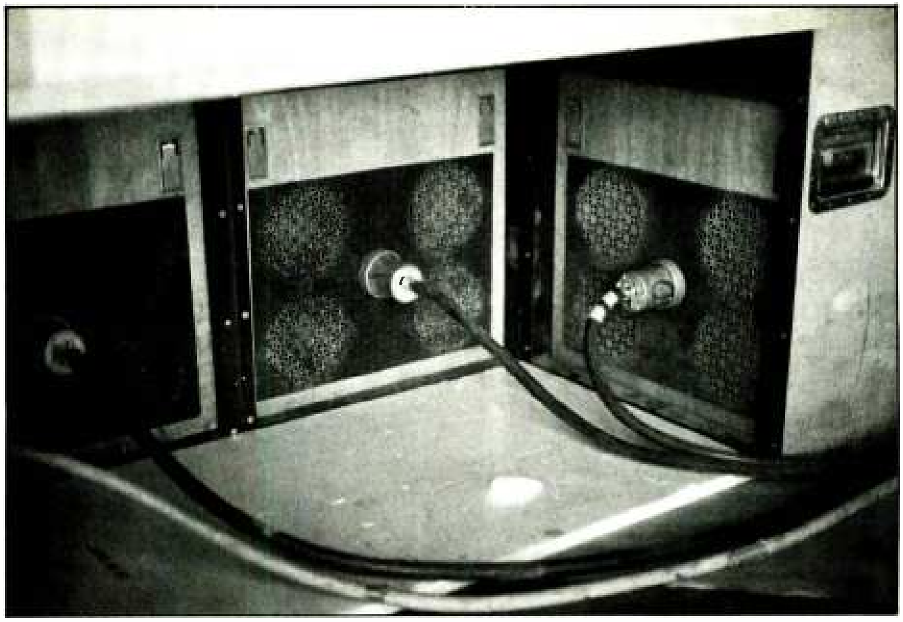


Figure 4: Clair house amp racks were positioned in tight groups of four. Intake and exhaust cooling fans were covered by a wooden enclosure, underneath which was placed a styrofoam pad that held piles of dry ice for added cooling protection.

The ice was delivered up to each tier by means of a makeshift elevator shaft built into the scaffolding: a chain-motor hoist was secured to a steel I-beam, and served as the motive force to raise and lower speaker cabinets, personnel . . . and dry ice.

To "power up" such a large number of amplifiers, Clair engineer Rex Ray found it easiest to leave all of the amps' power switches turned on, and rely on circuit breakers to switch the amplifier racks on and off. "I leave all of the amps on at night when we are ready to shut down," he explains. "We kill the racks a few at a time by switching off circuit breakers, and I stay by the breaker box for about five minutes until all the capacitors have discharged, then I kill the juice at the main disconnect switch. Then, in the morning, I reverse the procedure, and I listen to the 'thump' each time to make sure that every circuit goes live.

"Once, the entire system got turned on at once and everyone thought there had been an explosion! The sound of 360 18-inch speakers thumping at one time was something a lot of folks had never heard ... it echoed through the valley for what seemed like days!"

Press releases reported the US Festival system's audio power rating as being 400,000 watts. The Clair house

Figure 5: "Laredo" — a delay tower, Texas-style. One of four Showco speaker stacks, each tower housing 32 Arena System cabinets, and eight amp racks.

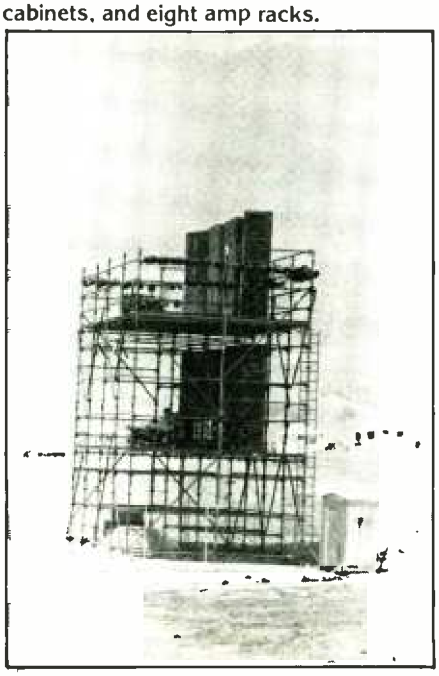
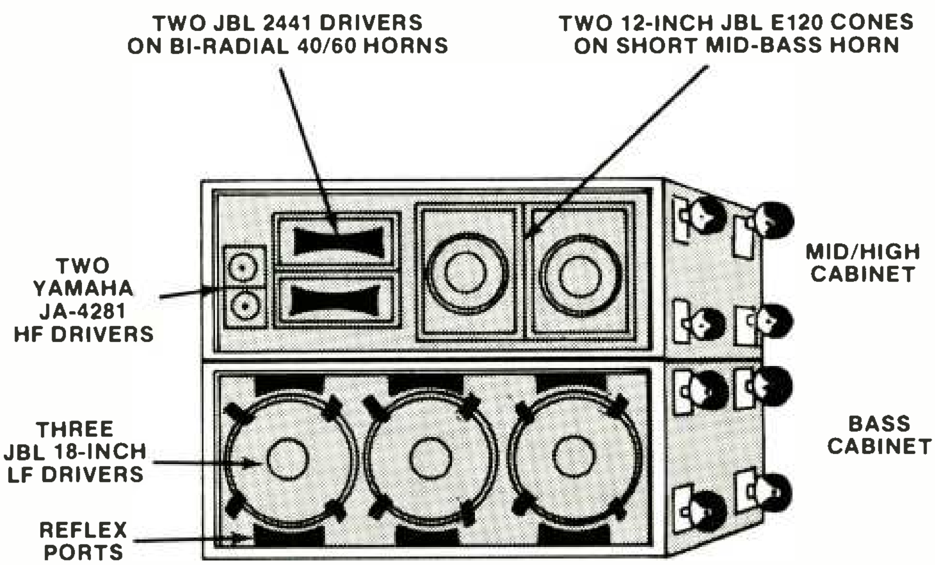


Figure 6: Showco Arena system stack: two identically-sized cabinets utilizing JBL and Yamaha components.

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system's 52 racks, the monitor system's 8 racks and the Showco delay towers' total of 32 racks gave a combined count of 92 amplifier racks. With eight channels per rack, this would divide out to an average of approximately 543 watts per channel — a not unlikely figure, considering the hefty Phase Linear 700-Bs, Crown PSA-2s and SAE 2600s in use during the proceedings.

AC Power Distribution

The 1982 US Festival relied entirely upon truck-mounted generators for its power source. For 1983, special power lines were brought in from a mile away, and which had been connected to the local Consolidated Edison grid. As emergency backup, a V-16 Caterpillar diesel was constantly idling, ready to drive a one million-watt generator set.

Clair's Rex Ray reports that his company's system had 800-amp, three-phase service available. "We are drawing about 425 to 540 amps per leg out of the available 800 — occasional peaks are seen to 500 during the loudest acts," he says. "We also see occasional peaks of over 500 amps on the neutral. Our line-voltage monitoring meter has been showing a very steady 123 volts, so we have pretty good current stability."

The stage gear received its own separate power distribution system, as did the Showco delay towers. To minimize hum problems, the lighting system was powered entirely by generator. Showlite's Ed Wanabo explains why: "We used on-site power generation for the two-megawatt lighting system to make sure that there was complete isolation between the lights and the sound system . . . separate AC source, separate grounding, everything. This way, there is no potential at all for line-carried noise interference — there's just no chance for the two grounds to loop between each other. Another common problem with a lighting system this big is RF interference; we solve that by modifying all of our dimmer racks in the shop before they ever even hit the road. Ours are much quieter than the stock units, and we rarely get any sort of complaints from sound companies about RF problems."

To ease logistical difficulties, the lighting system came in a full week and a half before the sound equipment. Overhead trusses were hung and wired before the sound trucks ever arrived. Additional equipment included 26 spotlights, 13 television cameras (several hung on tracks above the stage and remote-controlled), and two huge 40- by 60-foot Eidophor TV screens flanking the stage on each side. A 20- by 30-foot Diamond Vision screen hung above the stage was bright enough to be seen in strong sunlight; the two Eidophor screens were used only at night. Each of the Festival's 34 performing acts agreed contractually to be videotaped for a potential movie version of the US Festival.

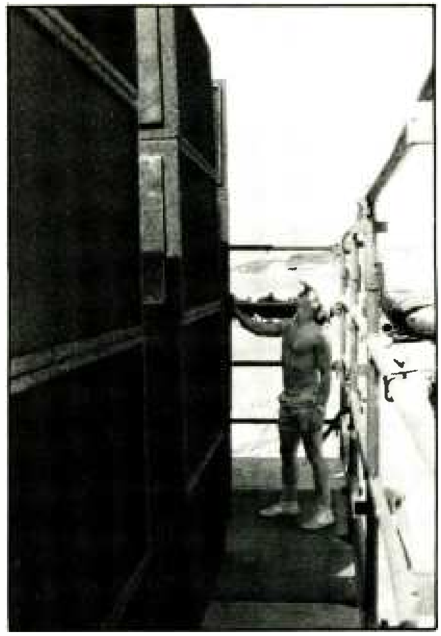


Figure 7: Showco speaker cabinets atop the "Dallas/Ft. Worth" delay tower, nearly 40 feet above the crowd.

Showco Delay Towers

With music fans in the rear of the seating area able to see the performers from a great distance away due to the large-screen projection units, delayed-sound towers were required to synchronize the audio with the video. Showco of Dallas, Texas, was contracted to provide additional sound reinforcement gear for four separate towers, which were situated approximately one-eighth of a mile out from the stage (Figure 5). With typical Texas humor, the Showco crew claimed the 54-acre audience area as property of the sovereign state of Texas, and named the four towers after four cities in Texas: Dallas/Fort Worth, Tyler, Lubbock, and Laredo; all cables running to and from these towers were accordingly labeled in this manner.

Each delay tower was essentially its own little self-contained fortress, a 1,600-square-foot area enclosed in eight-foot high chain-link fence. On the middle level of each tower was situated a Showco engineer who monitored the power amplifier levels, adjusted delay times, and maintained communication with the show's production chiefs via a headset-type communications system. All AC power lines, system-drive leads, and intercom connections were run through an underground conduit system that had been laid in trenches. These were covered over with earth months prior to the event, which gave the grass a chance to grow over the disturbed area. (A portable toilet facility also was provided in each enclosure for the engineer's convenience, while on the other side of the fence from it were thousands of persons who had to trek a quarter-mile back to the public facilities.)

The Showco speaker system consisted of the new two-box, four-way Arena system (AS) loudspeaker enclosures. Two identically-sized, 250-pound cabinets comprise one set of AS speakers (Figure 6). The bass cabinet contains three 18-inch JBL cones, while the mid/high cabinet houses two JBL E120 12-inch units, two JBL 2441 compression drivers mounted on 40 by 60 Bi-Radial plastic horns, and two Yamaha JA-4281 Super-Tweets HF units.

Each of the four towers was stacked with 32 cabinets: 16 bass boxes, and 16 mid/high units. Cabinets were stacked in four columns of eight, on two different tiers using a construction crane. The top cabinets were nearly 40 feet above ground level, and a step ladder provided access to the upper levels (Figure 7).

All power amps were set on the middle level (along with the engineer's ice chest) for easy access. Showco amplifier racks housed Crown DC-300-As and PSA-2s, each of the eight amp racks per tower driving two pair of AS speaker cabinets (Figure 8). The bottom amp in each rack was bridged into 4 ohms, making approximately 1,200 watts RMS available to a pair of AS bass cabinets. (The Arena System is essentially the old Showco Pyramid system boxes, which have been extensively refurbished and re-engineered, then fitted with the latest in transducers).

Delay Tower Electronics

Each delay tower was equipped with its own electronics rack, assembled specifically for this event. A Showco four-way electronic crossover split the signal being sent to each tower from the Clair house mix position, and an MXR third-octave graphic equalizer gave the tower engineer control over each speaker stack. Delay tower EQ settings were essentially flat, with gently-curved, 6 dB peaks introduced at the one-octave regions centered around 100 Hz and 5 kHz.

A roving engineer on an ATC (All-Terrain Cycle) three-wheeler patrolled the audience area perimeter and relayed necessary delay and equalization-change information to the production command post via walkie-talkie. This data was then passed on to the appropriate tower by hard-wired communications line. ("Dallas-Fort Worth! Kick up your mid-bass just a touch . . . wait a minute . . . more . . . dandy!")

For the one-eighth mile distance, a delay setting of approximately 600 to 700 milliseconds was required. Three of the towers were equipped with a DeltaLab ADM-310 digital delay unit (claimed by the manufacturer to have a frequency response from 20 Hz to 20 kHz, flat within +1, -3 dB). The fourth tower was supplied with a Lexicon Prime Time digital delay.

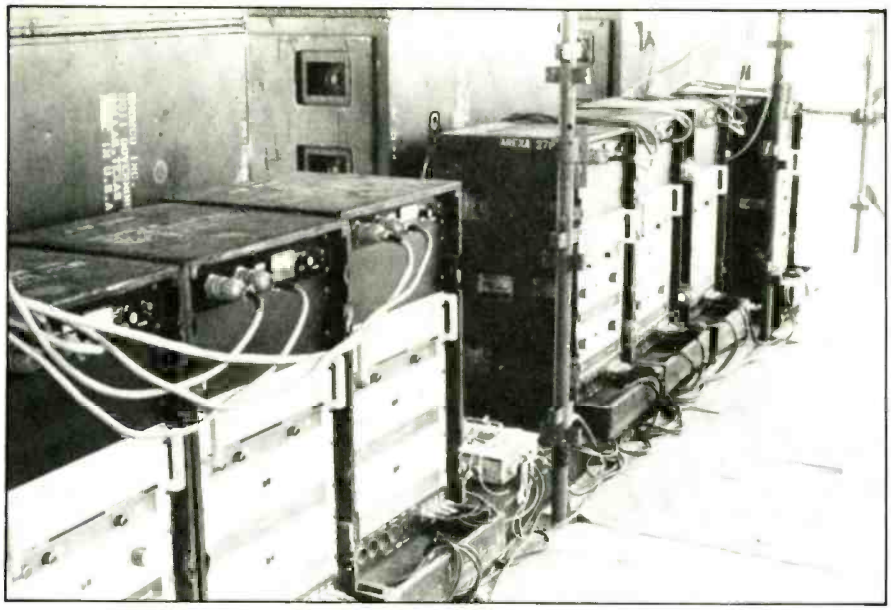


Figure 8: Showco amp racks for the delay towers: Crown DC-300-A and PSA-2 units, eight racks per tower.

House Mix Position

Situated only 150 feet out from the downstage edge of the performing area, the house mixing platform measured 24 by 40 feet. Surrounded by chain-link fencing, the platform served the lighting and special effects crews as well as the sound engineers (Figure 9). Clair supplied three of its 40-channel folding consoles. Two desks facing the stage were used for alternating acts; as one engineer mixed his band's set on one console, the second was being set up for the next act. A third board situated behind the engineers served to send out the five main output mixes: left stack;

right stack; spare left and right; and an auxiliary drive rack mix. This last mix went to a group of dbx Model 900 limiters set at unity gain, and used as line drivers for eight more isolated mixes. These latter included the four delay tower sends, a left and right mix to the video truck as a reference/backup source, and two more sound reinforcement mixes that went to the auxiliary centerfill cabinets, and the backstage priority guest seating area. (These two mixes were passed through Clair R-4 boxes, which basically are half-sized S-4s with fewer components). Figure 10 details the output mix flow.

To record the US Festival on multitrack for posterity and profit, Westwood One's 45-foot remote trailer was present on site during the proceedings. [For a full rundown of the mobile vehicle's facilities, see the Spring 1983 issue of *Audio Production for Broadcast — Ed.*] Westwood's Dave Faragher explains what the truck did with splits from the on-stage mike inputs: "We are laying down multitrack as a reference for future sweetened mixes, with crowd sounds picked up by audience mikes. At the same time, we are doing a mono broadcast mix, which we feed to the video truck next door. The Green, Crowe and Co. truck then uses our audio mix for the videotapes. Additionally, there is a satellite feed for a live video broadcast to Moscow over the Cable News Network." (The two mobiles are shown in Figures 11 and 12.)

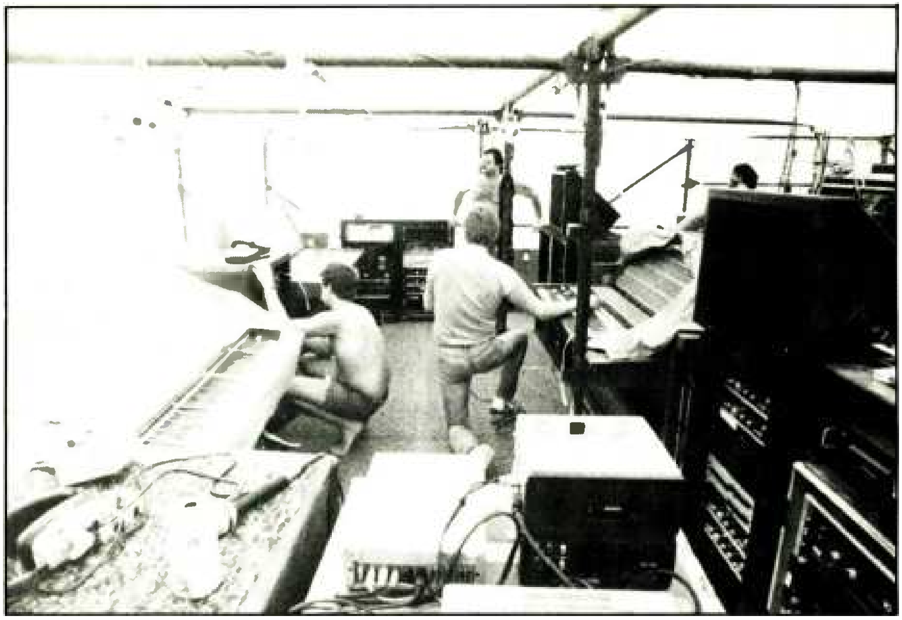


Figure 9: Audio section of the spacious 24- by 40-foot house mixing riser, with three Clair consoles. L to R: Clair engineers Dave Kob, Rex Ray, and Bruce Jackson.

Clair House Electronics

From the primary Clair mix output console, program feeds for left and right house stacks passed through Clair four-way electronic crossovers. Output signals from the crossover then hit a rack of 20 dbx Model 162 compressor-limiters, which served as line drivers, monitoring meters, and speaker protection for the various tiers of S-4 speaker cabinets. Each separate low, mid and high feed passed through a 162 unit before traveling down the output snakes to the amplifier racks onstage. As previously stated, other output mixes not requiring a Clair crossover split were passed through a rack of dbx Model 900 compressor-limiters.

System equalization was achieved by using White third-octave filter sets, two of these units processing the main left and right mixes. System frequency response was observed graphically on a Klark-Teknik DN60 real-time analyzer, which input was fed with an AKG C451 condenser mike positioned directly in front of the mixing area. Pink noise and pre-recorded music were fed through the system for an entire day as the various tier levels were set. (Pre-recorded music input to the system came from a Sony PCM-F1 Digital Audio Processor!)

Special effects and signal processing devices available to mixing engineers at the Clair position included an Eventide H949 Harmonizer, Lexicon 224 digital reverb, Lexicon Super Prime-Time DDL, and Orban Parasound devices for channel-patching on vocals. Each of the two identical mix consoles that saw alternating use were provided with similar effects racks.

Showco House Mix Position

In addition to Clair's three consoles, the house mix platform contained a Showco 30-input "superboard," which was used to mix the performance sets

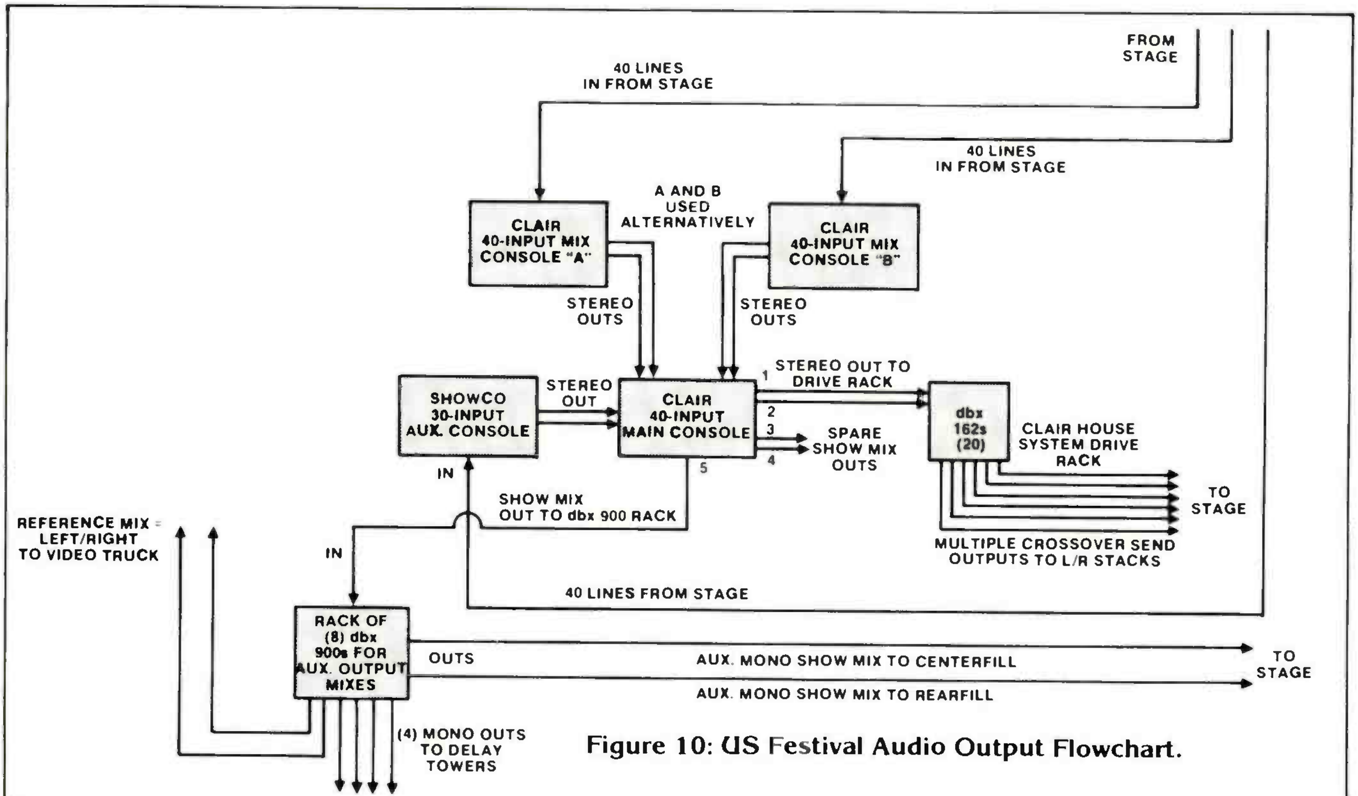


Figure 10: US Festival Audio Output Flowchart.

given by Van Halen and David Bowie, two of Showco's current touring accounts. A separate 40-pair snake cable carried stage input lines out to the Showco board. (All lines running from the stage to the house mix position were passed through a 4-foot diameter underground conduit with manhole covers and step ladders.) Showco engineers were able to do a mix on their own console for two of the show's most prominent acts, stereo outputs from this board being sent directly to the Clair main show console; a separate on-stage monitoring system was provided by Showco for these same two acts.

The Showco house mix station was equipped with dbx Model 900 compressor-limiters for individual channel patching, MXR 31-band graphic equalizers, a Klark-Teknik DN60 real-time analyzer, and Lexicon digital delay devices. Showco engineer Roy Snyder cited console familiarity and his company's clients' requests as two reasons for the separate console setup (Figure 13). The mixing position was tied into the same communications line that served the Clair engineers, as well as the Showco stage line.

Communications Systems

With over 30 engineers and technicians onsite directly involved with audio production for the US Festival — 12 from Clair, eight from Showco, four from Westwood One, and so on — a multichannel communications line was considered essential. Clair engineers ran their own Clear-Com line, as well as another separate intercom system that used an amp/speaker combination at both the house mix and monitor mix positions for a positive talkback device. Showco engineers had inter-console communications of their own.

The master production-line intercom system was provided under separate contract by Audiotechniques of Los Angeles. A Clear-Com SB-412 unit gave each station a four-by-12 matrix access (12 stations; four channels). This system carried signals for the lighting and sound crews, as well as an overall technical production line for UNUSON personnel, and a private special effects line to enable video crew personnel to give instructions to the special effects crews (lasers, etc.). Each remote truck *also* had its own wireless and hard-wired system serving relevant personnel. Walkie-talkie devices were used by roving production crew chiefs, and event security personnel.

Onstage Monitor System

Stage monitoring facilities for nearly every act appearing on the US Festival bill were provided by Clair Bros., the exceptions being Van Halen and David Bowie, which were handled by a separate Showco monitor system. Clair's engineers endeavored to provide each band's monitor mix engineer with the type of system with which he was familiar. Clair engineer Bob Weibel explains how he approached the complex task of providing stage monitors for over 30 different acts: "We figured that the best way to go would be to give each guy what he asked for. We put up the mikes he wants. We bring them up on the board in the order he wants. We patch in the extra EQ and compression he wants. That way each engineer is more comfortable when he finally gets right down to mixing the monitors. The less he has on his mind about figuring out the equipment the better."

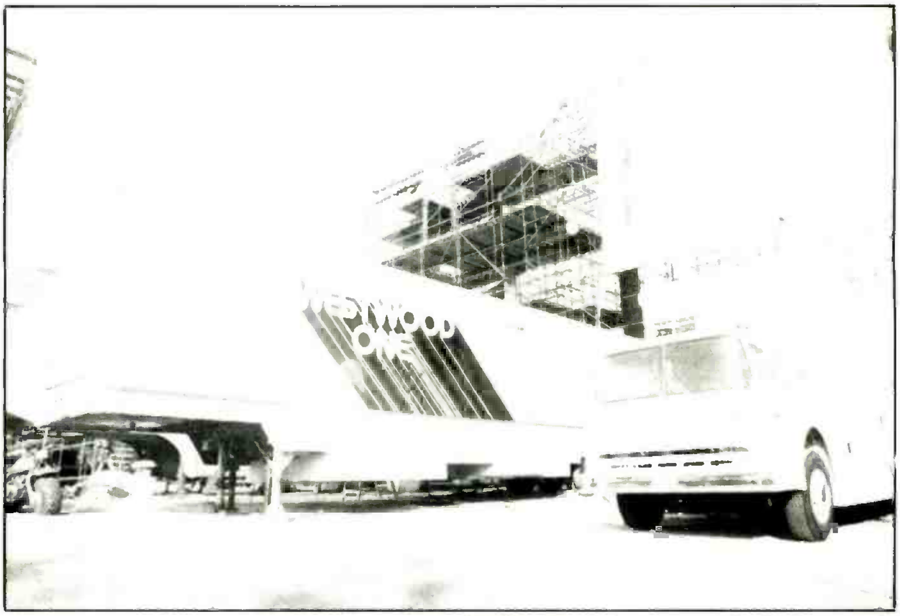


Fig. 11 (Top): Westwood One's mobile truck laid down multitrack for a future reference mix, and supplied the video truck with a broadcast mix. Fig. 12 (Right): Green, Crowe and Co. truck videotaped the Festival, and provided a satellite video feed for a live CNN broadcast to Moscow.

Weibel and the rest of the Clair stage crew had approximately one hour to do each changeover between acts, and he felt that only one set of monitor gear was necessary. "In the house, they have two sets of everything. Here, we are right on top of it all, and the change goes much more quickly," he offers. "Besides, our space up here is more limited, and two sets of monitor boards and EQ racks just wouldn't work out as well. UNUSON officials have given us a stage chart and input list for every act well ahead of time, and I am trying to meet personally with each monitor mix engineer to make sure that we know what he wants. Advance communications make it all go much more smoothly."

The Clair monitor system was based around a Midas 24-channel desk, an additional Midas 24 being available for auxiliary monitor inputs (Figure 14). Weibel felt that the Midas was a good choice of board for the US Festival: "I would be very surprised if most of the

Figure 13: Showco's auxiliary 30-input "Superboard" console was used for that company's two touring accounts performing at the US Festival, and had been positioned above and behind the Clair main consoles.

Figure 14: Clair senior monitor engineer Bob Weibel consults input lists for his two Midas monitor mix consoles. Eight monitor mix outputs were supplied, plus stereo sidefills.

guys who will be mixing these bands are not already familiar with the Midas. It's a pretty straight-ahead console, and has been around a good while; nothing too fancy.

"I am set up now with the two boards paralleled, but as I look at the list of acts we are serving, it looks as if no one will need the extra console."

Weibel felt that being prepared for unusual last-minute requests would ease his job as overall stage supervisor for Clair's system: "I have brought in more of everything," he explains. "I have over 50 floor slants available; we'll obviously never use that many, even if they all get blown up twice! And the extra Midas desk . . . I'd rather be able to do a bus parallel, and keep all eight mixes, than just have a small submixer. Plus, this gives me a complete spare desk right here in case I need it. Here, we are giving these acts everything from soup to nuts . . . it has to be all there.

"It *is* a lot of gear . . . it's a far cry from the days back when I was in college, and we used to just lay a column [speaker] in front of the band for a monitor and say, 'Here, guys, knock yourself out! Now, if one of these people needs compressors on all his vocal channels and that sort of thing, I just reach for a patch cable."

Monitor Electronics

The 10 mix outputs from the two paralleled Midas desks were fed into dbx Model 162 compressors. White third-octave filter sets and/or Klark-Teknik graphic equalizers were easily patchable into each output line, as per a given engineer's preference. A Clair electronic crossover split the bi-amplified mixes at

1.2 kHz, and lows and highs were pushed by left and right channels of specially modified SAE Model 2600 power amplifiers (Figure 15).

Clair engineer Mike Wolpert, who usually is found touring with Kenny Rogers' system, explains the monitor system in further detail (Figure 16).

"Our floor slants are set up with an internal passive crossover," he says. "That way, we can always get a good sound without even using a crossover. By changing a simple jumper, the boxes can be run two-way. For the US Festival, we have all bi-amped mixes set up. The floor slants contain a single 15-inch cone, and a TAD 1602 driver mounted on a specially-built wooden radial horn.

"With patch cables, I can quickly change any mix output from one area of the stage to another. I have extra dbx limiters available, extra graphics, even a Lexicon delay unit. Pretty much whatever they ask for, I can patch in immediately."

Clair's sidefill monitor stack gave every appearance of being able to provide adequate coverage for the 56- by 80-foot stage; each side contained no less than 16 15-inch speakers. Four cabinets per side each housed four JBL speakers, and was referred to as a 4145 box (Figure 17). Sidefill mids and highs were handled by Community Light & Sound's new M-4 diaphragm, and JBL 2441s.

"The M-4 is a 6-inch aluminum diaphragm with no cone at all," Wolpert explains. "Just a huge diaphragm surrounded by styrofoam. They work great outdoors . . . they sound really natural on this stage. We considered overhead monitors for a while, but decided against it due to the large, hard flat stage surface with no carpet on it."

Power for the huge sidefill stacks came from Clair's compact four-in-one "Boat Anchor" amp rack (Figure 18). "We call it that because it is so heavy for its size," Wolpert remarks. "It consists of four SAE 1,000-watt amps, put into one case along with cooling fans and an electronic crossover. You just run a mike cable to it with a mix output signal, plug it in, and you have instant sidefills. Here, we've put the Boat Anchor on a moving dolly right behind the sidefill stack, so the whole thing rolls around if needed."

Together, the two large sidefill stacks put out such an intense sound-pressure level that complaints were yelled out from all parts of the 436-foot wide staging area as one band's monitor engineer checked his centerstage vocal mike with a deafening "Check . . . One! Two!"


Figure 15: Racks of SAE 2600 amps, specially modified by Clair, to power the bi-amplified monitor mixes.


Figure 16: Clair engineer Mike Wolpert at the monitor mix position, stage left, with Klark-Teknik graphics and White filter sets available for monitor mix EQ.

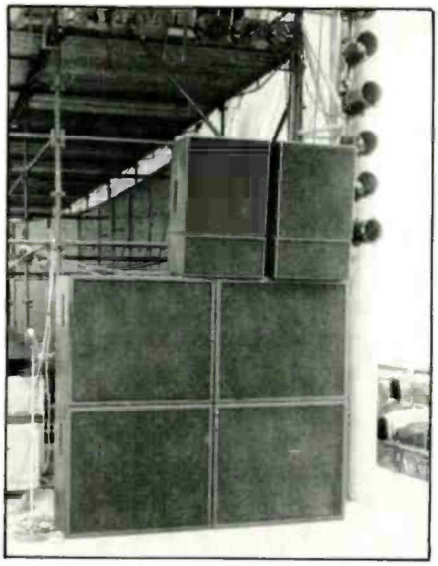


Figure 17: Clair sidefill stacks: 16, 15-inch cones per side, with Community Light & Sound's M-4 diaphragms on the mids, and JBL 2441s on top.

For heavy-metal drummers, Clair engineers had prepared a special monitor rig: four Crown PSA-2 power amplifiers built into a small rolling rack, which also housed an electronic crossover set at 250 Hz and 1.2 kHz. When wheeled into position by the drum riser, this rack powered four Clair R-4 boxes (housing the same '18s as the S-4 house cabinets), providing a very portable, very loud drum monitor set-up which needed, once again, only a mike cable for input once it was positioned on stage using its rolling riser.

Stage Miking

With as many as 10 bands appearing on the same stage in one day, microphone cables almost seemed to be flying through the air of their own volition. Applying a method to this madness was Clair engineer Kathy Sander, whose job it was to make *sure* that input line #19 was kick drum for Wall of Voodoo, for example (Figure 19).

"Just about whatever mike an engineer requests on a given instrument, that's what I have the guys put up there," Kathy comments. "Standard for vocal mikes are Shure SM-58s or Electro-Voice DS-35s, and that's what we'll place unless requested otherwise. As far as the drum kit, I have Beyer M88s, Shure SM-56s and Sennheiser MD 421s for toms. For kick, it could be an AKG D-12E, an E-V RE-20 . . . whatever. For overheads, lots of AKG C451s, Sennheiser 409s; lots to pick from. And I have plenty of SM-57s for everything else."

Kathy explained that each stage riser area was miked up on 6- or 12-line satellite stage boxes, and run into the main splitter on stage left. "We have two 40-line trunks going out to the house, so we can pre-set the rolling risers with mikes and cables, then make a quick-patch with the multiconnector into the correct trunk line. That way, we are already working behind the scenes on the next group's set, while the current act is performing."

In just five minutes, Kathy claimed that a major act could go from risers rolled into position to her crew being ready for mike check. "Normally, it would take about a half-hour to be that ready," she points out. "Our set changes are going a lot faster than that. Then, we have about 30 minutes to check lines in the house, the monitors, and at the recording truck."

While Kathy personally checked input lines for correct numerical designation and handled the master stage chart list, a crew of four Clair employees set mike stands and ran cables, moved floor slants, and placed microphones.

Mixing The Show

A total of 34 different performing acts means probably 34 different house mix engineers, and 34 different monitor mix engineers. To make certain that these individuals' needs were taken care of in the house, Clair engineers Mike Wolfe, Bruce Jackson, Dave Kob, and Mike Stahl took turns minding shop.

"Basically, when an engineer is mixing his act out here, he can do whatever he wants," Jackson remarks. "We've not set ourselves up as some sort of 'God Committee' to dictate how they will do their job, but they have access to all the gear that is here, provided they can show us that they know what to do with it.

"We have set the mix position at what is relatively the same distance away from the stage as these guys would be in an arena, which is where most of these people are used to mixing their acts.

"Our biggest problem, you could say, comes from guys who are here to mix a new act, and have never used such a large system as this before. It sort of intimidates them, and they get so hyper that we spend most of their set trying to just calm them down so they can work."

Jackson points out that no single individual made all of the decisions concerning the huge system's setup and operation. "We all talked it out beforehand, and discussed different methods and strategies," he recalls. "Then we each took a walk around the venue, and sort of got the levels and EQ set ahead of time, communicating back and forth with the walkie-talkies. The evening before the first day's performance, we had a local band do a live set to get a final system checkout, and for the video crews to adjust their cameras and things. Finally, The Clash did what you might call a dress rehearsal late that night, so we were actually quite prepared when it came right down to it."

The "Clair Committee Method" seems to have worked out quite well. Mike Wolfe was actually project manager, but system responsibilities were spread amongst several individuals. During the event itself, comments concerning the sound system were invariably positive.

HARRISON SM-5 MONITOR CONSOLE
A Cooperative Design Effort between Showco, Clair, and Harrison Engineers

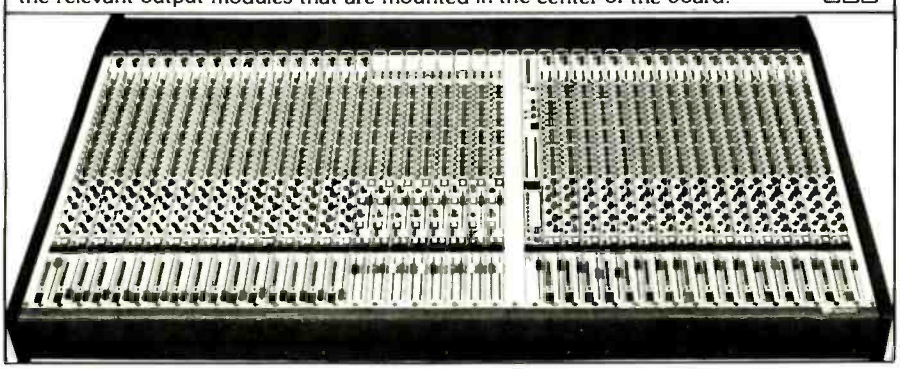
Perhaps the most interesting piece of hardware in use at the recent US Festival was the new Harrison 32-channel SM-5 monitor console. Introduced in June of this year, the SM-5 is probably the most complex console ever to be designed and built for the sound-reinforcement market. It is one of the few purpose-built monitor mixers available, and was designed from the ground up specifically to provide on-stage monitoring facilities. And, perhaps most importantly, the SM-5 represents a concert sound "first": both Clair and Showco engineers collaborated on layout and design specifications for the SM-5, and then presented the concept to Harrison for manufacturing. (In the past, both companies have either relied on commercially available products, or designed and built their own consoles.)

The US Festival has the distinction of being the first concert sound project in the United States to use the SM-5. Showco provided a stage monitoring system for two of the company's major clients, Van Halen and David Bowie, and which was completely separate from the Clair monitor set-up that served all other acts on the bill.

First conceived in mid-1982, the SM-5 went from sketched-out plans and wishes to reality in less than a year. A total of 16 boards initially were constructed by Harrison. Showco took delivery of 10 SM-5s in May of this year, and Clair ordered six boards. Clair's first console was shipped immediately to Europe for use with Rod Stewart's tour; Showco's first console was in use on-stage at the US Festival. Showco plans to use the board for all future and upcoming tours, including Linda Ronstadt, Peter Gabriel, and Rick James. According to Showco owner/partner Jack Maxson, "We've made the decision to concentrate our energies on doing shows. We'll let Harrison concentrate on manufacturing."

The SM-5 weighs approximately 300 pounds, and measures 40 by 71 inches wide by 11 inches high at the back panel. An internal carbon-steel frame houses 32 input modules, and ensures structural integrity; additional inputs can be added via an extender frame. "This console is really tough," comments Clair engineer Rex Ray. "It doesn't have any flex to it. You could probably run over it with a truck, and not hurt the frame. Four people can lift it easily, and the thing looks like it should last forever."

Other features of the SM-5 include 16 main monitor mixing busses; 16 group re-assign busses; four-band, full-parametric equalization on each input channel; a group muting matrix; and VCA subgrouping on all 16 output busses. Output LED metering is built into the relevant output modules that are mounted in the center of the board. □□□



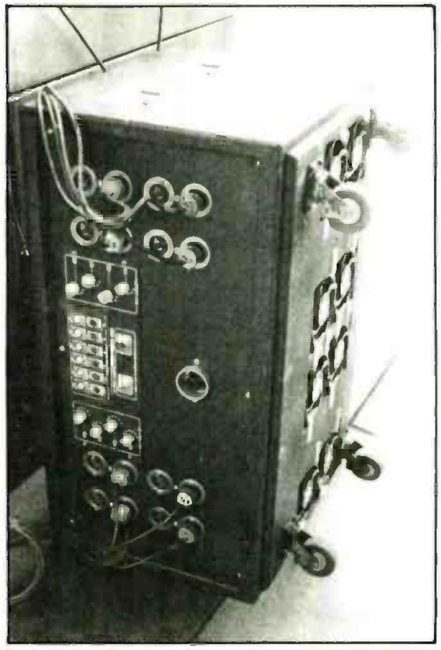


Figure 18: Clair's "Boat Anchor" amp rack: four 1,000-watt power amps in a compact box on wheels, complete with crossover and cooling fans for use with sidefill stacks.

System User Comments

Michael George, house mix engineer for The Clash, found the quality of the system to be excellent: "It was a *very* good-sounding system throughout the entire audience area. However, I felt that the volume left a little to be desired. I did not get an SPI meter reading, but it was nowhere near what I expected it to be in terms of level. As a matter of fact, in my conversations with the engineers for Men At Work and The English Beat, it came out that they felt the same way."

Clair engineer Rex Ray explains George's comments as follows: "The lower tier of our speaker stacks was greatly reduced in level compared to the upper tiers. At the console, you have to realize that a lot of the system's acoustical energy is actually going right over your head, where it should be, so that it is hitting the rear of the audience area. Looking at that huge pile of speakers, it might be a bit disconcerting to not be blown away by the sound, but the audience heard it just fine."

Another engineer pleased with the system's quality was Showco's Roy Snyder, who mixes house for Van Halen. "The cooperation between the two companies here has been excellent. We have a bit of friendly competition, but that's healthy."

Synder remarked that, for him, outdoor shows were nothing new. "I did the Rolling Stones outdoor shows last year. However, this is something unusual for Van Halen. We almost always tour indoor venues . . . they are really an arena band. For me, as a rule, mixing outdoors is easier, since you don't have walls and things to clutter up the sound with a lot of echo. Outdoors, the sound has someplace to go . . . you don't get a lot of standing waves building up. It is

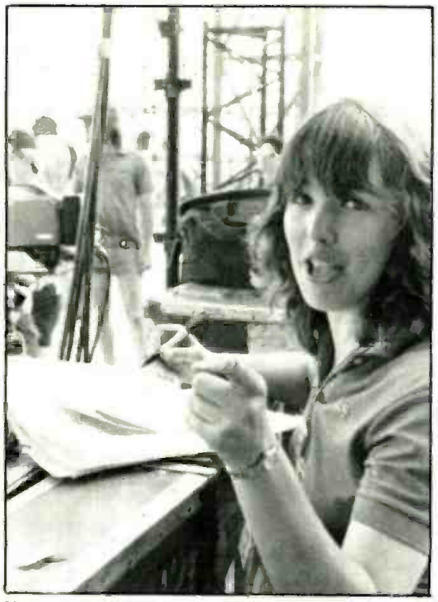


Figure 19: "Don't get me mad, or you won't have any vocals!" Clair engineer Kathy Sander was in charge of which mike line went where. Each act's engineers consulted with her prior to their sets, to ensure trouble-free set changes.

easier to hear what your system is doing.

"On the other side of the coin, there are unique problems. Wind conditions can really mess you up. You get a mix all set, then the wind shifts and you start hearing everything different, and have to change levels, EQ, everything. But here, we have one heck of a system, and I, for one, am enjoying myself."

* * *

If Woodstock started it, did the US Festival end it? Promoters are split on the issue. Can large outdoor rock festivals still be produced economically as they were in the Sixties?

Today's crowds are more sophisticated and demanding, and less likely to accept the poor-quality audio which characterized outdoor shows 15 years ago. Problems such as crowd violence, ticket counterfeiting, and prohibitive local regulations make it more and more difficult to predict a successful outdoor festival date. UNUSON's Steve Wozniak remarked in advance that, if the 1983 event lost money like the 1982 US Festival, this might be his last. (He reportedly lost $10 million on last year's Labor Day extravaganza.)

The 1983 US Festival was a showcase for some of today's hottest recording acts (demanding hot fees: reportedly, Bowie and The Clash each successfully negotiated a $1.5 million fee). The latest in audio and video hardware was on display. To some members of the audience, the entire event must have seemed like one gigantic live television show, with huge viewing screens on which to see the performers, who appeared as miniscule puppets from the rear of the seating area compared to the larger-than-life video image.

As an outdoor exercise in concert logistics and engineering, the US Festival was practically flawless. Hundreds of working personnel, millions of dollars' worth of sound and lighting gear, and dozens of trucks were organized and orchestrated into a momentous event.

Whether or not such overstated production events will be seen in the future, Showco's Jack Maxson feels that, for some parties, the US Festival will ultimately be very profitable. "Right now, some of our equipment suppliers are already starting to approach us for endorsements . . . you know, 'The amplifier which rocked the US Festival' . . . that sort of thing. We are having to be very careful about who uses our name in that respect, and the names of our clients in connection with us. However, ultimately, the equipment manufacturers can see that there is a lot of mileage to be gained from events like this one. The 'glamour' PA companies, the names which get linked with major artists . . . we all stand to benefit by that sort of advertisement, I think. As new products are developed for concert sound use by the manufacturers, companies such as Showco and Clair, along with many others, try them out. And an event such as the US Festival is one place to find out . . . does your system *work* . . . or doesn't it?" ■■■

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SOUTH STREET VENTURE

A Multimedia Program Highlighting New York City's Redevelopment of the Manhattan Waterfront

Complex Slide and Film Presentations with Timecode Synchronized Multitrack Audio

by Robert Carr

Renewing decayed urban areas and restoring historical sites are not new phenomena. But the latest craze in big-city redevelopment is a composite of both approaches. Boston tried it first with its harbor area, and the enthusiastic reception that the refurbishment received was enough to inspire New York City to try the same approach with its waterfront in lower Manhattan.

A highlight of the Big Apple's new face is an audio/video production entitled *South Street Venture*, sponsored by Norwalk, Connecticut-based Trans-Lux Corporation. Trans-Lux contracted Rusty Russell Projections to write the show and oversee the actual production. Having already completed a number of similar programs, including *Where's Boston?* (a salute to the Boston Bi-centennial in 1976), Smithsonian's *Nation of Nations* (Washington, D.C.), and John Hancock's *60th Floor Observatory* (Boston), Russell and crew were able to draw upon their experience, and incorporate the latest advancements in audio and video technology to present what promises to be the epitome of sight and sound presentations.

"Promises to be" is an appropriate phrase, because this article is being prepared prior to the program's scheduled opening sometime toward the end of September, 1983. [The production also forms part of the Social and Cultural Program organized by the AES, a special trip having been arranged for Monday October 10, during the forthcoming Audio Engineering Society Convention, New York City — *Editor.*]

The best approach to understanding the various audio aspects of the 59-minute, 55-second *South Street Venture* program is to be provided with an overview of the show itself. Upon entering the theater, the audience will see in front of them a 65-foot screen extending across the front of the room. The projection surface arcs slightly at both ends and merges effortlessly with the condensed, but proportionate, models of local buildings, comprising a scenery set that runs to the back of the theater along the two side walls. During the approximately four minutes allotted for the audience to enter and find their seats, the audio system plays an introduction of music, sound effects, and background voices. "The idea is to prepare the audience for the program," says soundtrack supervising engineer Thom Foley, "and get them used to being surrounded by the show."

Synchronized Film and Video

The visual portion of *South Street Venture* is a skillful marriage of 64 Kodak slide projectors and three motion-picture projectors, focused not only on the front screen, but also on various secondary screens throughout the auditorium. (Thirty-two slide machines cover the front, and the remaining 32 are scattered around the house for special effects.) The primary film projector, a 16mm Elmo containing the image and voice of the principal narrator, Stephen D. Newman, one of the stars of *Sophie's Choice*, is mounted on a lazy-susan-type of device that swivels back and forth by computer control to give the illusion that Newman is walking (or running) across the entire screen, or doing entrances and exits on and off the "stage" area. Synchronized with his motion is his narrative audio track (a monaural, optical film soundtrack) that is panned among five custom Altel speakers mounted behind the screen.

With the aid of a second, unseen narrator, actress Colleen Dewhurst, Newman guides the audience through a tour of the neighborhood, and explains the historical background and significance of particular local landmarks. As he talks, points at sites of interest, and directs the action, slide presentations of various configurations join him on stage — sometimes at the opposite end of the screen, and sometimes covering him completely until he disappears from view.

Once in a while, the narrator decides to stroll down behind the sets that cover the side wall. (The sound of his footsteps and the conversations he has with other unseen, but overheard, residents play through the appropriate speakers mounted in the scenery. In total there are 18 TOA Model RS-21M, and two Yamaha S3115H monitors for sound effects: four under the audience, three above, six behind one wall, and seven behind the other. Once he's reached the correct window, the shutters pop open to expose a latex mask covered mannequin on which Newman's talking head is projected via a second projector; the monitor speaker behind supplies his voice.

Other special house effects (there are just over 100 in the entire show) include little sight and sound gags, like a window appearing to burst into flames, and a machine in the front of the house that creates a light sea fog. Other visual and sonic effects include a specially prepared section of ship's rigging that *almost* falls on the audience, plus smoke, lightning, and cannon flashes. Kintek sub-woofers support the bone-shaking low frequencies that accompany explosions, and occasionally nasty weather.

The show concludes with a four-minute fiesta sequence, which features background samba music and credits on screen while the audience exits the auditorium. The audio for the entire presentation is played back by two, eight-track Otari MX-5050B MkIII-8 half-inch machines. While one show runs, the other copy is being rewound. At 59 minutes and 55 seconds, the timing allows five seconds between tapes, which is sufficient to switch from one machine to the other, and always start the next showing on time. In addition, if one machine ever breaks down, the second eight-track provides an instant backup.

Obviously, a show of this complexity comprised several major production

A scheduled visit to South Street Venture has been arranged as part of the Social and Cultural Program for the forthcoming AES Convention, New York Hilton, October 8 thru 12.

steps — separate sessions to record the music, sound effects, and narration; and to sequence, mix, and ensure that everything lined up when the project was assembled. Without becoming too bogged down in details, the production schedule can be broken down into separate areas, as follows:

Sound Effects

Aside from a few pre-recorded original sound sources pulled from a personal collection, engineer Mary Jane Soule was assigned the responsibility of creating or recording live all of the sound effects for the project. Her choice of machine was a two-track Nagra IV-S, as it has been for previous projects she's worked on for Rusty Russell. She then edited and dubbed the resultant tapes via a dbx noise-reduction unit to another Nagra half-track running at 7½ IPS. The result was a library of more than 60, seven-inch reels of assembled and cataloged sound effects recorded at appropriate levels.

Narration

Narrators Stephen D. Newman and Colleen Dewhurst cut their voice tracks at F-V Sound, a New York City narration room, with Vernon Sollecito engineering. Dewhurst recorded her final version at this point, because she is never seen during the show. However, Newman acts as the on-screen host, and more care was required to prepare his final tracks. This initial session produced only a scratch track of his narrative. Soule then edited and assembled both narrations into order on the same reel for dubbing at a later time. She also put together a transfer of all spoken parts (principal narrations and character voices), cut them into sections with the approximate timings, and laid down guide tones before each segment. Those were then sent to San Francisco for use as guide tracks to complete the music recording.

Music Tracks

An historical presentation such as *South Street Venture* demands that extensive research be completed to find the appropriate period pieces to enhance every scene. The arranger/conductor chosen for the task was Ed Bogus, a San Francisco-based music director known for his soundtracks behind *Fritz the Cat* and the *Garfield* movies. Bogus' favorite Bay Area studio is Russian Hill, a 24-track room that became the location for music recording. Engineer Richard Greene used conventional miking techniques to capture the music live on an Ampex MM-1200 24-track, and a custom Helios console; Dolby-A noise reduction was used on all tracks throughout the entire audio production. Thom Foley arrived for the last day and a half of recording, and assisted Greene with the mixing.

— Supervising engineer Thom Foley —

Recording the music for *Venture* was similar to any soundtrack session for a motion picture. The major difference, however, becomes apparent when one realizes that there was actually no film to use as a guide, as would normally be the case on a typical scoring stage. Instead, Soule's previously prepared voice transfer was laid down on track #1 of the 24-track. Bogus, who had already composed the music to a copy of those same edited narratives, then, in turn, could listen to reference track as he was conducting the musicians to record the soundtrack.

"A minute-and-a-half segment of narration may be divided into three sections," Foley explains. "We would record the music tracks for the first section on tracks 1 thru 12; the music for the second section on tracks 13 thru 24; and switch back to 1 thru 12 for the last section. When we mixed them together, we could maintain the illusion of continuity by shifting back and forth between the two sets of 12 channels."

Mixing was done to discrete four-track assignments on a half-inch Ampex AG-440, the recording once again encoded with Dolby-A noise reduction. Four monitor speakers were placed in front of the console to simulate the playback layout of the theater where the show was eventually going to play. All the tracks were sequenced and numbered for future reference.

Making It All Fit Together

Once the music was in the can, the tough part really began — assembling the various components into one final master audio reel for use in the show. Thom Foley and Rusty Russell flew back to the East Coast, and set up shop at Northlake Studios in White Plains, New York. Basic equipment included an MCI JH-114 24-track, and an MCI JH-632 console with automation, the latter facility considered essential for the project's rigorous mixing demands.

The work schedule ran from 6 a.m. until between 3 to 6 p.m. every evening. "Because we were working in a rock studio that also did heavy-metal sessions every night, the [apparent] levels on the tape machine would change drastically from one day to the next," remembers Foley. "I had to align the 24-track *every* morning, set up a half-track machine for the sound effects, a four-track for the music, and a mono machine to play back the narration." (Remember, both Dewhurst's final version and Newman's "ghost" track are on the same reel.)

Starting with the clean two-inch tape, approximately four-and-a-half minutes of click track was laid down on channel #1 for reference. (The low level of the signal eliminated the need for a buffer track.) The click track served as the timing guide for the walk-in section, which would hold the background sound effects that entertain the audience as they are finding their seats. "We like to finish the audio for the whole show," Foley says. "And then, knowing what we've done, find new sound effects and background voices that are reflective of the overall mood of the presentation to assemble the walk-in."

At the four-minute mark, the actual

Audio Equipment for South Street Venture Presentation

Narrator Stephen Newman's optical voice track plays through a Kintek optical pre-amp Model KT-41; a dbx Model 140, Type II noise reduction unit; a Yamaha Model 1027 one-third octave graphic EQ; an Arion 895 decoder, which pans the sound across five custom-built speaker cabinets, supplied by Altel Sound, and powered by five channels of Yamaha PM2100 amplifiers. Elmo projector carrying Newman's visual presentation is controlled by a Projector Control system manufactured by Northern Precision Labs.

Each of the four principal audio tracks are played through a Dolby-A decoder; and a Yamaha Model 1027 third-octave graphic EQ; a Yamaha 5115HT speaker cabinet powered by one channel of a Yamaha PM2200 amplifier. Bass reinforcement for extreme low frequencies is handled by a Kintek KT-90 sub-woofer system, with a built-in, 500-watt Kintek amplifier.

Each of the two effects tracks runs through a Dolby-A decoder; one channel of a Yamaha PM-2100 stereo amplifier; a Yamaha Model 1027 one-third octave graphic EQ; and an Arion 895 decoder that assigns speaker routing to any combination of the 18 TOA Model RS-21M and two Yamaha S3115H effects cabinets. □□□

TRACK ASSIGNMENTS FOR MASTER 24-TRACK MUSIC, EFFECTS, AND DIALOG TAPE

Track #1	— MCI automation data and/or click-track
#2	— Narration: Newman guide track
#3	— Character Voices
#4	— Narration: Dewhurst, plus character voices
#5	— Character Voices
#6	— Character Voices
Tracks 7 thru 10	— Music A
Tracks 11 thru 14	— Music B
Tracks 15 thru 23	— Sound Effects
Track #24	— MCI automation data

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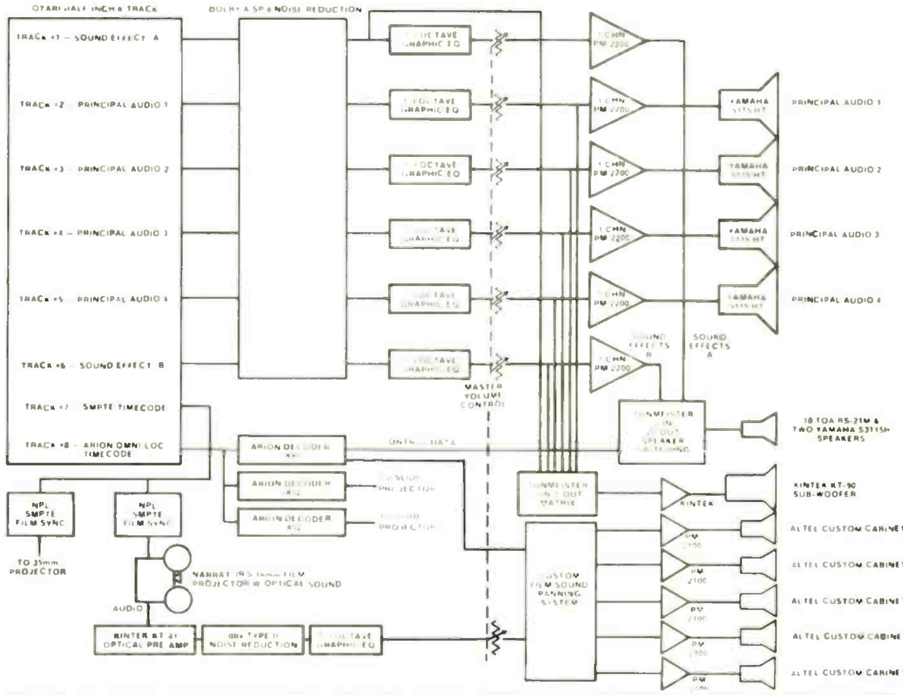
show begins. Newman's ghost narrative was recorded on channel #2 of the 24-track, and used as a guide only. [See accompanying sidebar for additional track assignments.] No SMPTE timecode references were used for this portion of the mixing. Following the schedule outlined in a production book put together by Russell, the various audio elements were added when needed and mixed using the MCI console automation.

"We could mix only about a minute of soundtrack at a time," Foley remembers, "because the panning was so intricate. It took three of us more than 100 hours to cut and paste the complete hour soundtrack. The first pass we'd mix just the voices; the second pass would put the music with the composite voices. The third pass we'd mix and pan the effects, and the fourth time through we'd update whatever was needed."

The result was two master reels of eight-track, one-inch tape recorded at 15 IPS with Dolby-A noise reduction. Those two reels were, in turn, transferred to two reels of half-inch tape at 7½ IPS, which were spliced together and coded with 24-frame SMPTE on track #7. The final Dolby-encoded, eight-track arrangement was as follows:

Track #1 — Sound effects "A"
Track #2 — Principal audio (music, background voices)
Track #3 — Principal audio
Track #4 — Principal audio



Track #5 — Principal audio
Track #6 — Sound effects "B"
Track #7 — 24 FPS SMPTE timecode
Track #8 — Arion Corporation "Omni-Loc" timecode

Timecode Reference Points

Tracks 1 and 6 are individually switchable to any one of the 20 effects loudspeakers. The principal audio tracks are tied to four Yamaha speaker cabinets located in the front of the theater — one track per cabinet. At first, Stephen Newman's "ghost" vocal found its way on to track #8 of the half-inch tape. "We needed Newman's voice and SMPTE as references for editing the final visuals. Once that was taken care of, we erased his rough narration and replaced it with the 'Omni-Loc' timecode that would sync the soundtrack to the Arion decoders."

Under the technical direction of Tim Burnham, three stock Arion decoders were installed to handle the switching cues. One Arion Model 895 was responsible for operating Newman's projector; the switching for all of the sound effects through the TOA and Yamaha speakers, and the low-frequencies through the sub-woofers (when necessary); and some of the house effects. Two Arion Model 852 units run the 64 slide projectors, the remaining house effects, and movie projectors. Programming of the system was done on an IBM Personal Computer.

Of the three motion-picture projectors in the set-up, only two were controlled via SMPTE timecode: Newman's 16mm, and the 35mm machine that provided the additional moving sequences, such as aerial shots, throughout the film. The third projector, a 16mm with no soundtrack, and responsible for the "talking head" that appears in a window of the scenery, was simply turned on and off by the Arion for the short segment it furnished.

Of particular interest is the manner in which film-sync sound engineer John Hampton was able to add Newman's part to the show. To ensure synchronization between voice and visual, the narrative had to be laid down on the optical track of the film stock. Once the music and sound effects were mixed, the finished tape was sent back to New York City, where Newman recorded his final narration in real time against the master audio. For the film shoot, which was done at the same time as the narration recording, the master audio tape containing all the program elements and SMPTE code were resolved to a tape containing a 60 Hz sync tone. That tape was played back on a Nagra two-track, whose crystal-sync controlled the film camera being used to shoot Newman. Newman, in turn, wore a wireless receiver and earphone so he could hear the soundtrack minus his original scratch track. His live narration was recorded on yet another Nagra held in time with the sync tone from the first Nagra. After that, the transfer to 16mm optical with SMPTE was easy.

To increase the signal-to-noise ratio, and generally improve the sound quality of the optical track, Russell Projections employed a Kintek pre-amp KT-41 through a dbx Model 140, Type II noise reduction unit, and a Yamaha Model 1027 third-octave graphic EQ. Initial tests indicate that the system functions as predicted, and helps cut down unwanted hiss and noise from the optical soundtrack.

As mentioned earlier, *South Street Venture* promises to be the epitome of sight and sound presentations. We should be able to judge for ourselves during the Fall AES Show, and on visits to the Big Apple. □□□

PERFORMING MEANINGFUL NOISE LEVEL MEASUREMENTS IN THEORY AND PRACTICE

by Paul C. Buff, Technical Consultant
Valley People, Inc.

The specification and measurement of audio noise levels, and their relationship to signal and overload levels, are of utmost importance in the sound studio. Yet the engineer or technician who is prepared to accurately define and measure these parameters is indeed of a rare breed. Perhaps equally rare is commercially available equipment capable of performing the task.

Which Noise is the Real Noise?

By definition, noise is anything that is not signal. While we are concerned with *all* noise sources, if our measurements are to be of any value in analyzing or improving our equipment, we should be capable of separating the various sources of noise. In characterizing the various noise mechanisms we may encounter in audio equipment, it is important to note that there are two primary categories: true thermal noise, inescapably present in any system due to thermal agitation of the electrons themselves; and induced, or error noise, such as hum, oscillation, RF pickup, power supply noise, etc. When we are dealing with a complex system, consisting of multiple components or modules, it is likely that thermal noise is the predominant mechanism within the components or subassemblies, while induced error noise results from extraneous pickup in the system's wiring and interconnection. By applying logic and sensible measurement techniques, we should be able to determine what we are trying to measure, and what to do if improvements are to be made.

Thermal or "Gaussian" Noise

Whenever heat is applied to matter, a resultant agitation of electrons takes place. When that piece of material — be it a conductor, semiconductor or insulator — is a part of an electronic system, the result is random, but predictable noise. Without delving deeply into the physics of thermal noise, it is important that we understand its laws of behaviour. The most important aspect is that thermal noise, in a non-frequency-selective system, appears with equal energy per unit of bandwidth. For example, the same amount of thermal noise energy exists in the frequency band from 100 Hz to 110 Hz, as exists in the band from 10 kHz to 10.01 kHz; both bands are 10 Hz wide.

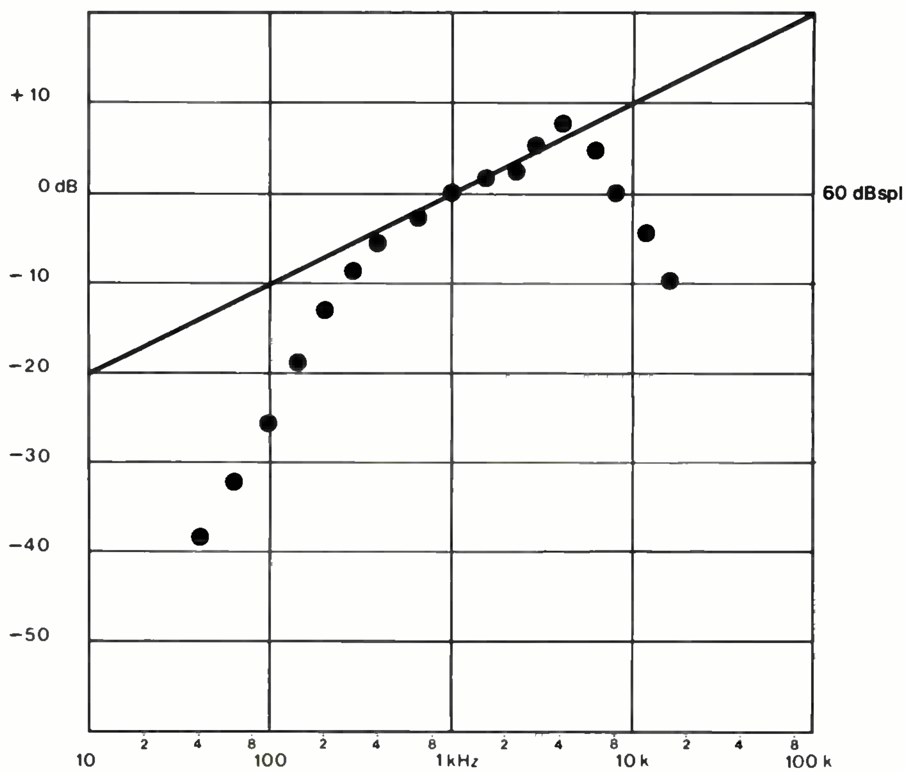
But this is unrelated to the nature of hearing and how sounds and music are created, since we relate what we hear to "octaves." The band of, say, 100 Hz to 200 Hz is given more or less equal importance as is the band of 5 kHz to 10 kHz. While each of these frequency bands forms an octave (or doubling of frequency), the number of cycles contained in the lower octave is only 100, while 5,000 cycles are contained in the higher octave. Thus, while the musical, or aural weight, of the two octave examples are more or less equal, we find the thermal noise energy contained in the 5 to 10 kHz octave to be some 50 times as great as that in the 100 to 200 Hz octave. This easily explains why pure thermal noise, or "white noise," sounds "hissy," or accentuated at higher frequencies, as shown in Figure 1.

If we take the accepted 10-octave range of hearing to be one extending from 20 Hz to 20 kHz, and divide it in the aural middle, we find five lower octaves from 20 Hz to 640 Hz, and 5 upper octaves from 640 Hz to 20 kHz. In calculating the spectral content of thermal noise within this same range, we find 31/32 of the total noise in the upper five octaves and only 1/32 in the lower five.

The point to be made here is that when examining pure thermal noise in a relatively flat response system, the entire lower half of the audio spectrum may be filtered out in order to remove error noise, such as hum and power supply ripple from the measurement. The result of using, for example, a 400 Hz high-pass filter for this purpose is the introduction of a negligible error when measuring thermal noise.

Taking the case a bit further, assume we have a piece of equipment having flat response to 500 kHz, thus designed for excellent transient and high-frequency phase response. Now, suppose we measure the residual noise level of this equipment with a wide-band meter exhibiting equal or greater bandwidth. Because of the nature of the thermal noise produced in the equipment, we will find that 25 times as much noise power is present in the ultrasonic region above 20 kHz, as is contained in the audible portion of the spectrum. Such a measurement, then, is purely *academic*, and has little bearing on how much noise we will *hear* when listening to the device. Thus, while the ultrasonic

Figure One: Spectral Density and Loudness of Thermal Noise. Solid Line represents spectral density of thermal noise with 0 dB at 1 kHz, while the dotted line represents relative loudness of thermal noise in dB SPL at a moderate listening level.



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region may be important to the transparency and transient response of the system, we have no business measuring its noise content if your objective is to determine how "quiet" the system will sound.

20 Hz to 20 kHz "Flat" Noise Measurements

The basic premise of measuring audio noise levels on a flat, or unweighted basis, is to restrict the measurement bandwidth to the range of human hearing. Hence, most manufacturers will specify an unweighted noise measurement band of from 20 Hz to 20 kHz. Many engineers and technicians make the mistake of fitting a pair of passive RC filters with -3 dB points at 20 Hz and 20 kHz to a wideband meter, expecting to produce the desired measurement bandwidth. Because of the gentle 6 dB per octave rolloff of such simple filters, a considerable amount of noise energy present above and below the desired band may be measured. The equivalent noise bandwidth is considerably greater than that calculated.

In looking at Table 1, it can be seen that the actual filter -3 dB points required to simulate a 20 Hz to 20 kHz equivalent noise bandwidth are a function of the number of poles contained in the filters. For simple 6 dB per octave filters (single-pole), the correct frequencies are 31.5 Hz and 12.7 kHz, respectively. As the number of poles is increased, the -3 dB points of the filters approach the absolute cutoff points that would be obtained with infinite pole filters. In practice, the sharper the filters, the greater the accuracy of obtained readings on equipment of various ultrasonic bandwidths. In a practical sense, equipment having 18 to 24 dB per octave filters, at the appropriate -3 dB points, will yield the required accuracy for confident field readings.

If the measurement equipment has a selectable 400 Hz or similar multipole highpass filter, measurements may be made with and without the lower audio octaves, to determine if the noise observed is primarily pure thermal noise, or comes from power-line related sources.

Use Your Eyes and Ears

This author believes the most fundamental mistake made in field measurements of noise is to simply connect a meter, read the answer, and believe it. The most versatile measurement equipment available is with you at all times —your *ears*. Besides holding your head together, the ears are very useful in determining the nature of what is being measured. While a noise meter will tell you the total amount of noise within the selected band, it will not readily tell you whether it is thermal noise, hum, leakage, or what have you. By always monitoring the signal under measurement, with the monitor gain set way up, your ears will instantly tell you what the meter cannot. If hum, buzz, leakage, or oscillation is inducing the error, you will hear it. If the system sounds quiet, but measures noisy, trust your ears — you must have a ground loop or other flaw in the meter connection.

Similarly, your eyes can prove very valuable. All serious noise measurements should be made with a good oscilliscope connected along with the meter, so you can see the waveforms being measured. If there is hum, or RF, or oscillation present, it is pretty hard to miss on the 'scope screen. In too many cases, an engineer or technician will connect a meter to a measurement point, without suitable filters and without aural and visual monitoring, and come up with readings 20 or 30 dB in error. In most cases, the error is caused by hum, RF, and ground loops in the meter connection, thus making the whole process a waste of time.

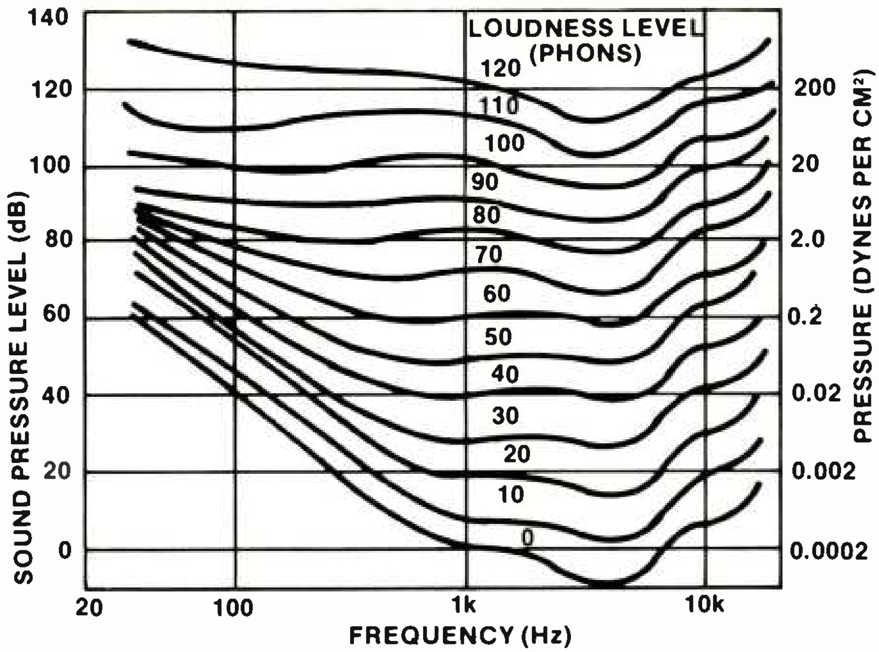


Figure Two: Fletcher-Munson Curves (Taken from "Audio Cyclopedia"; Courtesy, Acoustical Society of America.)

Weighted Noise Measurements

Any number of studies have been conducted over the years to determine the nature of human hearing. Typical of the results of these endeavors are the familiar Fletecher-Munson curves presented in Figure 2 (courtesy ASA — *The Audio Cyclopedia*). Here, it is shown that the characteristic of the average ear is a pronounced area of sensitivity in the 2 kHz to 6 kHz region, with reduced sensitivity at both the lower and higher frequencies. Since the degree of sensitivity loss at the spectrum extremes is a function of the sound pressure level being monitored, it is impossible to draw one universal curve of how the ear perceives the total frequency spectrum. Nevertheless, the spectrum is definitely not perceived on a uniform basis. Thus, if we are to create a means of noise measurement that accurately defines what the perceived noise level will actually be in listening, the non-linear response of the ear must be considered.

TABLE 1: FILTER -3 dB POINTS FOR EFFECTIVE 20 Hz to 20 kHz NOISE BANDWIDTH

Filter Slope	High Pass	Low Pass
6 dB per octave	31.4 Hz	12.7 kHz
12 dB per octave	25.0 Hz	15.9 kHz
18 dB per octave	23.2 Hz	17.2 kHz
24 dB per octave	22.4 Hz	17.9 kHz
30 dB per octave	21.9 Hz	18.3 kHz
36 dB per octave	21.6 Hz	18.6 kHz
Infinite dB per octave	20.0 Hz	20.0 kHz

In the discussion thus far, we have concerned ourselves with essentially unequalized equipment, such as recording consoles and peripherals, including limiters and other effects units. Equipment of this sort is characterized as producing essentially "white" or "Gaussian" thermal noise as the primary component, while low-frequency noise sources in such equipment usually are considered error noise. Since the noise spectrum of unequalized equipment is usually predominant in the upper octaves, the flat 20 Hz to 20 kHz form of measurement provides a reasonable basis of comparison, and relates well to perceived noise.

Whenever we are dealing with equipment that employs considerable equalization, such as found in tape, disk and other systems, the spectral nature of the noise can depart dramatically from the Gaussian, or equal-energy-per-cycle formula. Notably, tape and disk playback systems may employ up to 40 dB of low-frequency boost in the equalization curve, thus passing large amounts of low-frequency thermal noise, as well as becoming extremely sensitive to hum pickup. Because of the relative insensitivity of the ear to low and very high frequencies, the use of flat, or unweighted, measurement techniques on equipment employing large amounts of equalization leads to measured noise figures that relate poorly to the perceived noise.

In trying to produce a system of measurement that better correlates measured noise to perceived noise, various schemes have been introduced to "weight" the measurement curve in a fashion similar to the inherent weighting of the ear. In creating such a weighting curve, many factors have been considered, including a consensus of what a "typical" listening level might be, since the listening level has a profound effect on the required weighting. Most notable, and most universally accepted, of these attempts are the "A" weighting curve shown in Figure 3, and the CCIR-ARMS weighting curve of Figure 4. Of the two, the "A" curve is the older, and most often used in the United States, while the CCIR-ARMS curve perhaps represents research more directed toward current product design. (The CCIR-ARMS is a derivation of the rather complex original CCIR method, and is more adaptable to affordable instrumentation.) It is not the intent of the author to make a claim in favor of any of the methods commonly used for measurement, but to present those most necessary in the implementation of meaningful field measurements of today's equipment.

Noise Measurement Equipment

In addition to a good set of ears, a good oscilloscope, and a knowledge of the nature of noise, the serious noise analyst needs some form of metering equipment suitable for the task. Specifically, the metering equipment should be capable of accurately covering the decibel range anticipated and, ideally, should have a multipole 20 Hz to 20 kHz noise bandwidth position, a selectable multipole high-pass filter at about 400 Hz, and "A" and CCIR-ARMS filter positions. For reasons defined later, the equipment should have balanced differential signal inputs, and a buffered input monitor point to allow connection of unbalanced monitoring equipment. The inclusion of selectable peak, RMS, and averaging measurement capability can be beneficial, though not entirely necessary.

Such equipment generally is not available from "test equipment" manufacturers, since their product lines often are directed toward broader segments of the electronics industry than the specific professional audio market. "State-of-the-art" in the majority of test equipment lines is toward very accurate, digital-segmented display gear. While these units are very advanced and versatile types of equipment, they are, perhaps, the very *worst* thing to attempt to use in studio measurements. In addition to never containing the filters necessary for pro-audio noise determination, digital displays are nearly impossible to read in the ever-changing world of audio level measurement. While 0.1 dB resolution might be wonderful for the sake of esoterica, it becomes rather a joke when 5, 10, or 20 dB errors pop up due to unsuitability of the equipment for the task at hand.

Measurement Technique

That is where it really takes place. There is rarely a case in audio noise measurement where the inherent accuracy of the equipment is not a lesser factor in the final result than are the *techniques* employed. This caveat applies particularly when stages or components having very low inherent output noise are under measurement. This author would make the guess that in perhaps 80% of the cases involving measurement of stages or components having output noise less than -70 dBv*, considerably more technique-induced error noise appears on the measuring meter, than does actual equipment noise. In perhaps another 80% of the cases, the measurement bandwidth or weighting is probably unsuited to the task.

The most pervasive sources of error noise in the measurement of low-noise stages are to be found in one of the following categories:

1. Inclusion of ground loops in the measurement due to:
   A. Unbalanced metering inputs.
   B. Connection to inappropriate circuit points.
   C. Power-line AC interference caused by inappropriate connections.
2. Inclusion of RF interference due to:
   A. Unbalanced metering inputs.
   B. Connection to inappropriate circuit points.
   C. Inappropriate shielding in the metering leads.
   D. Inappropriate filters in the metering equipment.

Other common sources of measurement error are to be found in equipment or metering devices located near intense electromagnetic fields (such as power transformers); inappropriate grounding techniques in the installation itself; and inappropriate shielding of low-level stages. A potential source of a lot of headaches also can be thyristor light dimmers in the building, as well as fluorescent lights and certain office equipment. A newer problem that can be extremely hard to isolate is the presence, and particularly connection to, the ever growing array of digital equipment found in studios. When digital switching signals start getting on the ground-

*In this text, dBv is a voltage measurement referring to 0 dB at 0.775 VRMS.

Figure Three: "A" Weighting curve Per ANSI S-1.4-1971 and IEC 123, IEC 179.

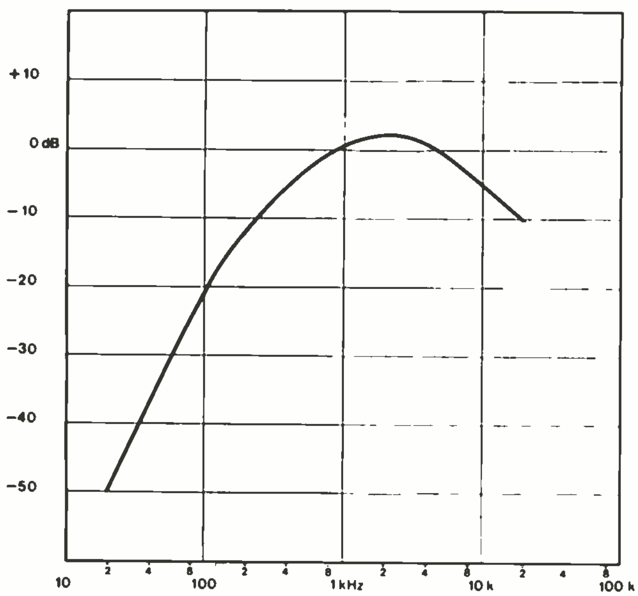
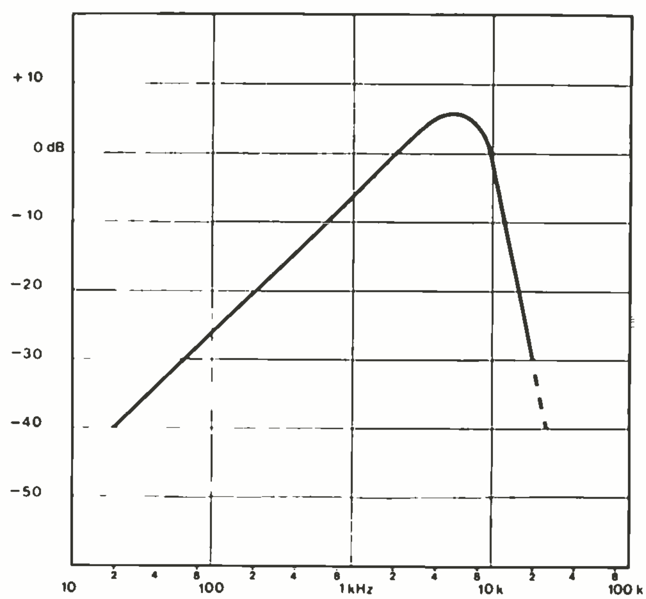


Figure Four: CCIR Weighting Per CCIR 468-2. Unity Gain at 2 kHz Per Dolby ARMS Proposal.



ing system (which they always do), they can become almost impossible to isolate.

In order to cope with this list of potential problems and error sources, the following general techniques are offered:

If the equipment to be measured is a peripheral or component that can be removed from the studio environment, do so. If possible, isolate the equipment on a test bench. The latter preferably should have an aluminum window screen covering the underside and, if possible, this screen should be grounded to earth. The case ground, or power supply ground, of the equipment under test also should be connected to the same earth ground. The test bench should have a good monitor amp and speaker(s) capable of good low-end response.

An oscilloscope, preferably with differential input capability, should be available. For maximum freedom from power-line induced error noise, it may be advised that an isolation transformer be used to feed all equipment, and that all case grounds, or power plug "third leads," be connected to a good earth-ground, rather than to the AC conduit ground. If an isolation transformer is unavailable, it is still a good idea to route case grounds to earth directly, rather than through conduit grounds.

The importance of excluding power-line grounds from the measurement loop cannot be stressed too much, since this usually is the primary source of error noise. To this end, it is strongly recommended that the only connection made to the output of the equipment under test be a bridging differential meter connection taken directly across the output terminals of the equipment being evaluated; such a practice precludes the parallel connection of monitor amps or oscilloscopes to the equipment output. In order to facilitate the connection of unbalanced monitoring equipment without introducing error loops, it is desirable to have a noise meter employing differential inputs and buffered monitor output. With this structure, unbalanced 'scopes, monitors, and similar devices may be connected to the meter's dedicated monitor output, such that monitoring can be accomplished without making unbalanced connections directly to the equipment under test.

Once the test set-up is in place, it is desirable to listen carefully at high monitor volume, letting your ears define the nature of the noise present. At the same time, the 'scope can be viewed for the presence of error noises, such as hum, buzz, or oscillation. If you hear, or see, anything besides *clean* thermal noise, you should try to determine the source. While it is possible that these error noises are emanating from within the equipment under test, it is, perhaps, more likely they are being induced from light dimmers, strong magnetic fields, or powerline ground loops. Only after you have become convinced that you are measuring *only* actual equipment noise, can you believe what the noise meter reads. Beyond this, the use of appropriate filters, monitors, and techniques will allow you to define the nature of the noise that actually emanates from the equipment. For example, if significant hum levels prevail, you won't form the opinion that the equipment is "noisy"; you will realize it has a hum problem. Corrective measures would then consist of looking for power-supply ripple, poor shielding, etc. By the same token, if your measurements disagree with the manufacturer's specifications, you will be able to make intelligent reports to the manufacturer with respect to the problem.

In-Studio Measurements

The proper techniques for measuring equipment hard-wired into an installation are much the same as those outlined above, but can be more difficult to control. The likelihood of substantial error signals induced through wiring and grounding flaws is rather high. Situations such as these dictate that a competent analysis be conducted to isolate real equipment noise from measurement- and wiring-induced noise. When making measurements of a module, or component, or sub-system within a console, for example, the following general procedure is recommended:

First, try to isolate the section under test as best you can; disconnect any connections to the section input where possible. If it is impossible to disconnect inputs, noise levels at the input should be metered and monitored to determine whether any noise, leakage, or other signals are being fed in. As with a test-bench measurement, to avoid introducing ground loops not normally present in the installation, it is most desirable to connect only differential input equipment to the section output. If a noise meter with a buffered input monitor jack is available, an oscilloscope and monitor amp should be connected to this point, so that you may see and hear exactly what the meter is reading. As before, listen and look for error noise, and experiment with system grounds and connection points to attempt to reduce or eliminate error noise. If you have digital equipment in the same room, you should suspect the probability of switching noises being induced into the system.

In console systems, it is certainly possible to encounter an assembly of components with excellent noise levels, yet a system that is intolerably noisy. By developing techniques that allow the causes and types of noise present to be pinpointed — as well as the signal flow through the system — you will then be prepared to explore corrective measures.

Signal/Noise/Headroom Considerations

Simply determining the residual output noise level of a piece of equipment will not tell you much about how quiet it will sound. You must also consider the typical signal levels, the overload or clipping point, and the amount of gain that follows the stage. In a system designed for optimum overall performance, the nominal signal levels are placed as close to the overload point as is practical, and as determined by the variability of input levels anticipated. Where the input signal is inherently confined to some nominal level, such as following a channel, effects send, or monitor fader, it is common to allow about 20 dB of "headroom" between the clipping point and the nominal signal level. This sort of headroom will allow the passage of highly transient signals at levels around 6 dB over the established norm without clipping. In stages where levels are not under tight control, such as between mike pre-amplifiers and faders, greater headroom is indicated to allow for the wider range of anticipated input signals.

It must be remembered that it is the difference between typical signal and residual noise that determines the actual noise performance of a system. Thus, for a stage having a given clipping point, the higher the nominal signal level, the greater the SNR (signal-to-noise ratio). Obviously, great care must be taken to allow sufficient headroom to accommodate the range of signals involved without the danger of transient clipping.

Specifications

Any useable specification of equipment noise performance must include, in some manner, the residual noise level, the recommended nominal signal level, and the overload point. Additionally, the specification must include the noise measurement bandwidth, or the type of weighting filter to be used. Specifications such as "80 dB SNR" essentially are meaningless. In equipment where the distortion characteristics are a function of signal level, it also is necessary to have a knowledge of just what the distortion/signal level relationship is, so that intelligent decisions may be made in the trade-off between low distortion and noise operation.

Noise Build-up in Multichannel Systems

If the engineer or technician is to make meaningful assessments of signals and noise in multichannel systems he or she must be conscious of how the various components build up within the system. Starting with signals, there is no direct simple relationship between levels at individual channels, and signal levels resulting from combined sources; it is all a function of the composition of the music on the various tracks. A broad rule of thumb might be that the total output signal may be around 6 dB higher than the typical signals at each individual channel.

The build-up of multiple noise signals, on the other hand, is very predictable. Specifically, random noise, such as thermal noise, builds up at the rate of 3 dB increase for each doubling of the number of channels summed. Thus, with respect to a single channel, two channels will result in a 3 dB noise increase, four channels is equal to +6 dB, eight channels +9 dB, 16 channels +12 dB, and so forth. When the noise source is coherent, rather than random, such as is the case in most hum pickup situations, the build-up is 6 dB for each doubling of channels. Thus, a noise level increase of 24 dB can result from the combination of 16 tracks, where coherent hum is the noise source.

In order to exemplify this relationship, assume we have a 32-channel mixing console with all inputs active, and being summed for a final mix. Assume each module has an output noise of -85 dBv, and a signal level of -2 dBv. If all channels are summed at unity gain, it is reasonable to predict a 6 dB addition in signal level, or a summed output of around +4 dBv. If the individual channel noise of -85 dBv is the result of pure thermal noise, the 32 noncoherent noise sources will sum to produce a 15 dB increase, a residual noise level of -70 dBv, and an output SNR of 74 dB. On the other hand, if the channel noise were predominantly coherent (in-phase) hum, the total summed noise would increase by 30 dB, to -55 dBv, for a resultant output SNR of only 59 dB. Thus, it is seen that the presence of coherent noise on individual channels of a multichannel system must be carefully analyzed.

dBv versus dBm

Prior to 1976, there was a prevalent trend in the professional audio industry to use incorrect terminology in the specification of audio voltage levels. Thanks to the willingness of *R-e/p* to open a controversy instigated by this author around that time [December 1977 issue; "Console Specifications" — *Ed.*], the majority of the industry has realigned its method of specification to correct the mistaken identity of the term "dBm." For those still unclear on the proper specification of audio voltage levels, the following will briefly explain the terminology involved.

. . . continued on opposite page —

VALLEY PEOPLE ADVANTAGE MODEL 310 AUDIO NOISE AND LEVEL METER

Technical Description and Operation Abstracted from Manufacturer's Literature

The Advantage Model 310 Audio Noise and Level Meter is described as the first low-cost, noise measurement device incorporating the specific features required for analyzing noise performance in modern audio equipment. The device consists of an expanded-scale meter that covers -100 to +30 dB (re: 0.775 VRMS) in two ranges, and specially designed selectable filter sets that allow measurement of noise in an equivalent 20 Hz to 20 kHz bandwidth, 400 Hz to 20 kHz bandwidth, weighted per ANSI S-1.4-1971 "A," or to CCIR 468-2 per the new Dolby ARMS recommendation. Meter responses are selectable for average, true RMS, and peak-responding characteristics. The unit is housed in a 5¼- by 8- by 8¼-inch (H×W×D) steel enclosure.

By using Trans-Amp™ technology for the metering inputs, the company claims to eliminate several sources of possible noise measurement error inherent in more expensive equipment. Inclusion of buffered access points allows connection of oscilloscopes, monitor amplifiers, or customized filter networks without affecting the accuracy of the measuring device. Use of logarithmic techniques in the basic circuit design, the manufacturer says, produces meter scaling that is accurate and easily readable over a 70 dB range, and provides sufficient dynamic range to allow level measurements to +30 dB, while retaining accuracy and essentially non-frequency-dependent response to less than -100 dB.

Both the high- and low-pass filter sections used for the unweighted measurements are designed to provide the correct indicated equivalent noise bandwidths, a feature which, the manufacturer says, enables greater accuracy in evaluating audible noise performance than has been previously available in more expensive, general-purpose instruments. ■■■

The term "dBm" was widely used in the days when systems commonly employed 600-ohm output impedances coupled to 600-ohm input impedances, for the maximum transfer of *power*, not voltage. The term is still widely used in systems where *power levels* are to be defined. Specifically, the term "0 dBm" indicates a *power level* of 1 milliwatt, and does not define voltage at all. Now, it happens that when 0.775 volts RMS appears across a 600-ohm load, the power dissipated in that load is 1 mW, thus 0 dBm is present. Since 600 ohm was the standard load impedance used in professional audio systems, *voltage* responsive meters labeled "dBm-600 ohm" often were used, and which produced a zero reading when 0.775 VRMS was applied. Now, while the meter actually measured a *voltage*, it did so to determine *power* level dissipated by the known 600-ohm load resistance.

As audio systems began to depart from the 600-ohm/600-ohm "power matched" concept and into mechanisms optimized for maximum voltage transfer (low-impedance outputs coupled with high-impedance inputs), the erroneous practice of using the term "dBm" to specify a voltage level began. For instance, if 0.775 VRMS appeared across, say, a 10 kohm load, it still was often called "0 dBm." Now, the amount of power present in this example is *not* 1 mW; it is only 0.06 mW. Thus, the *power* level is -12.2 dBm, not the "0 dBm" that would be read on a dBm meter.

Since most modern specifications of signal level — including noise specifications — represent voltage levels, the power term "dBm" is irrelevant, for the same reasons that you do not refer to your 120V house wiring as a "10,000 watt" line. The correct, and now common, method of specifying audio signal voltages is in "dBv, or dB 0.775 VRMS" — a voltage level relative to 0.775 volts RMS. When the term "dBv" appears alone, it is usually taken to imply a reference level of 0.775 VRMS, as is the case in this article. When the term "dBV" appears (capital "V"), it is the European terminology of referencing the voltage to 1 VRMS, as opposed to 0.775 VRMS.

Average, RMS and Peak

Most signal and noise measurements are taken with averaging-type meters; that is, the common VU meter. While calibrated to *display* the RMS value of applied waveforms, these meters do not actually measure RMS voltage; rather, they *imply* an RMS value based on the measured *average* value. The implied RMS value is only accurate when the signal is a steady-state sinewave. When non-sinusoidal waveforms are presented, such as music waveforms, RMS reading errors will develop, depending upon the waveform complexity and transient characteristics involved. For this reason, typical music signals contain significantly more RMS, or effective, energy than is indicated by the meter. Along with some error in the reading of RMS energy on complex waveforms, very serious discrepancies exist between the magnitude of waveform's instantaneous peaks, and what is indicated on VU-type meters. It is typical for waveform peaks to be in the region of 15 dB higher than the VU meter reading on complex, transient music, such as percussion.

True RMS meters are designed to accurately measure the effective energy of *steady-state* waveforms of varying complexity. Thus, a steady saxophone or violin note, for example, both of which produce the same sonic energy, will measure the same on a true RMS meter. However, non-steady state waveforms, such as percussion, will still contain considerably more energy than will be indicated on RMS meters. Additionally, there is still no correlation between what is read, and the magnitude of the waveform peaks, since complex waveforms can have a very high "spike" in the waveform, yet still yield low RMS or effective power.

Peak responsive meters, as their name suggests, specifically measure the value of waveform peaks, without regard to the RMS or average values. Accordingly, their readings will not indicate apparent loudness, but rather whether or not waveform peaks are approaching circuit clipping points.

Since system noise ordinarily is a steady-state phenomenon, noise-level measurements are most accurately performed by the use of true RMS metering. The error induced by reading noise levels with averaging meters, however, is small, amounting to 1.05 dB under-reading on thermal noise.

The use of peak responsive metering does have its purpose in the investigation of noise sources. For example, if noise spikes, such as might result from digital signal leakage or from lamp dimmer interference are present, extraordinarily high peak meter readings will indicate their existence.

* * *

From the content of this article it should now be obvious to the engineer or technician that no single combination of weighting filters and meter responses is universally suited to noise measurement. In order to correctly specify, identify, and analyze the various types of noise encountered in audio equipment, one must have access to measurement devices incorporating specialized features. The adage, "If it's worth doing, it's worth doing right," certainly applies to the investigation and measurement of noise in audio systems. The result of using inappropriate equipment and techniques is, all too frequently, an assemblage of meaningless data. Even worse, the data may then be used to "correct" equipment flaws that may not exist.

All photography by Beth Gwinn

TREASURE ISLE RECORDING

A Nashville Facility Featuring a Live Acoustic Treatment for Control Room and Studio Area

by Sam Borgerson

he final months of 1981 presented a bleak prospect for anybody planning to build a new ding facility. Recording budgets were following record sales on a downhill slide, while at the same time building costs were following interest rates up to new peaks. It was not the best of times — and it may have been the worst of times — to build a deluxe new studio from the ground up. The crunch was on.

Fred Vail and Dave Shipley had little choice, however; they had just been evicted from their former studio.

One year earlier, in August 1980, Vail and Shipley had opened a modest 24-track operation in leased space inside a large office building on Nashville's Music Row. Operating as Island Recorders, the small studio hummed with activity for the next year. Then word came down about the planned resurrection of a huge, unfinished studio immediately adjacent to Island in the same building. Bullet Recording, a state-of-the-art audio/video recording complex, was about to be born. [See October 1982 issue of *R-e/p* for a rundown of Bullet's design and construction — *Ed.*]

Island's proprietors, not wanting to be overshadowed by this dazzling new neighbor (and facing non-renewal of their year-to-year lease), reluctantly decided to take their fledgling recording business elsewhere. Island's lease was taken over by Bullet, and the small studio was swallowed up, to emerge later as Bullet's Studio B.

Although painful at the time, this unforeseen eviction gave Vail and Shipley the opportunity — albeit under difficult circumstances — to actively pursue a dream that had been incubating for several years: to build their own new studio, a facility unlike any other existing in or around Nashville.

Despite all the cards stacked against them, Vail and Shipley achieved their goal. When Treasure Isle opened in late September, 1982, it was immediately hailed as "one of a kind" among the dozens of studios now operating in Music City. Instead of a small, dead room, the style in vogue during the studio boom, Treasure Isle offers a very live 30- by 42-foot room with a 24-foot ceiling. Instead of a closet-sized drum booth, the studio features a booth the size of most apartment living rooms — except that most apartments don't have 13-foot ceilings. The vocal booth — tastefully furnished with couch, lamp, and artwork — could easily double as a miniature parlor. From the upstairs office and producer's hideaway, sliding glass doors open out on to a balcony that projects out into the studio. The spacious control room, again tastefully decorated, gives Nashville its first "LEDE-type"* mixing environment.

And, when clients tire of recording chores, they can relax in the brick and stained glass atrium, take a shower in the restroom, unwind in the whirlpool bath and sauna, work out on the exercise cycle, watch the color TV, mix a drink at the bar, cook up a snack in the microwave oven, or romp around in the public park and playground across the street.

Studio Origins

Although an impressive facility by any standard, Treasure Isle becomes downright surprising when you consider how it was financed and built. Don't look for any big-time corporate backing behind this studio. What's more, neither Vail nor Shipley can be found on the front of a hit album, and neither has written any smash country songs. Basically, Treasure Isle is the product of two dedicated (should we say desperate?) entrepreneurs who defied the economic odds, and proved in the process that ordinary people can still compete in the state-of-the-art studio game.

Beginning in Sacramento as a disk jockey and program director at a radio station in the late Fifties, Fred Vail followed a zig-zag path on his way to the studio business. While still in California he ventured into concert promotion, then worked as a producer and national sales manager for Teen-Age Fairs, a Filmways project. His long association with the Beach Boys, which continues to this day, included stints in personal management, as well as operation of the band's label, Brother Records. He also served as a promotion executive at RCA and Capitol, and worked independently as a promotion consultant for the Beach Boys, Waylon Jennings, RSO Records, and Eric Clapton.

Migrating to Nashville in 1974, Vail hooked up with Dave Shipley. A former studio guitarist, Shipley spent several years producing custom albums (primarily for gospel groups) before turning to engineering on a full time basis. Vail and Shipley soon formed their own production company, Mariner Productions, leasing the results to record labels. During these struggling years the pair discovered two essential facts about independent production: first, you make very little money if the record is not a hit; and, second, you can spend a *great* deal of money just paying for studio time.

"Owning our own studio seemed a legitimate, reliable way to have a steady

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*Live-End/Dead-End™ is a trademark of Synergetic Audio Concepts, San Juan Capistrano, California. Treasure Isle's control room was designed strictly according to LEDE principles by Richard Lee, who is licensed by Syn-Aud-Con for LEDE design. However, because Treasure Isle has not yet completed the full LEDE certification procedure, it is referred to as an "LEDE-type" control room — *SB.*


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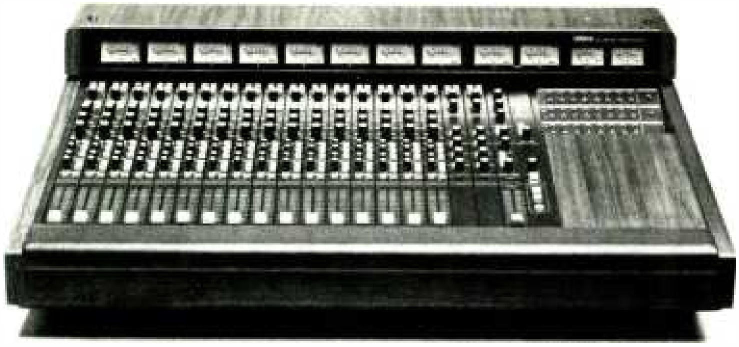

RM1608

SPECIFICATIONS

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

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

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

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

MAXIMUM VOLTAGE GAIN (INPUT GAIN "−60")

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

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

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

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

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

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

Hum and Noise are measured with a −6dB/octave filter at 12.47kHz; equivalent to a 20 kHz filter with infinite dB/octave attenuation.
*0dB is referenced to 0.775V RMS
• Sensitivity is the lowest level that will produce an output of −10dB (245mV), or the nominal output level when the unit is set to maximum gain
• All specifications subject to change without notice.

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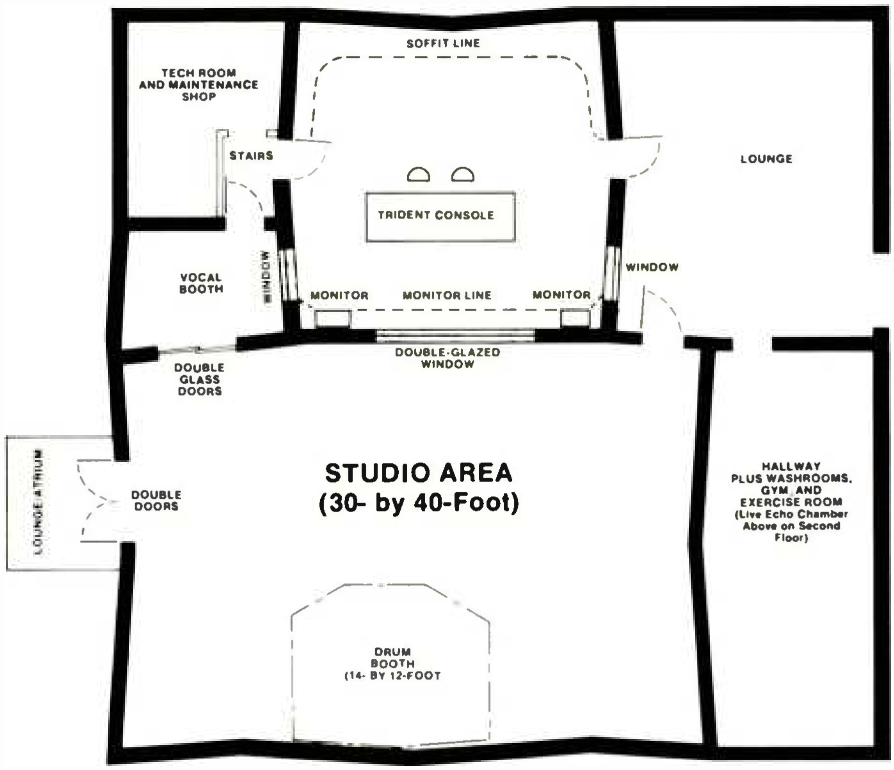
income while we still pursued our own production projects," Vail says. "So in 1979 we took an option on a piece of property in Berry Hill, a Nashville suburb. We though if we are real lucky, we'll build a small studio out there someday. If not, we could buy it as an investment, or convert it into an office.

"Then, in early 1980, we put some costs together for building a small but very high quality studio. A friend of mine who has a studio in Las Vegas, Jon Parks, said he would help me put together the financing. We spent several months working out the pro formas. We had costs; we had proposals; we even had letters of recommendation from clients who said they would use it. We also had estimates from architects, contractors, and equipment suppliers. We ended up with a 35-page document, but eventually we had to give up. It never came through."

A few months later, while engineering a project at Paul Richey's Richey House Studio, Shipley learned that the small 24-track operation was up for sale. The two partners, seizing what seemed to be a perfect opportunity, quickly put a proposal together to buy him out. Vail raised some money within his family, Shipley borrowed a modest sum on his '66 Mercedes, and they closed the deal. Richey already had sold his MCI 24-track before the papers were signed, so the new owners quickly went further in the hole with the purchase of a Studer A80VU multitrack. They assumed Richey's bank note on the remaining equipment, which included a Harrison console, and took over the year-to-year lease agreement offered by the building's owner, Financial Institution Services, Inc.



Of course Vail and Shipley were both aware of the cavernous, unfinished studio — vacant for the past six years — located next door. They simply assumed it would remain vacant for at least three more years, at which time they could once again consider building a new facility. At least that was the plan.

Less than one year later they were back out on the streets.

"First we considered all the alternatives," Vail continues. "We could buy an existing studio, but it would be less than what we really wanted to work in. No good studios were for sale. Or we could go into an existing building, and again put our money into a leaseholding arrangement, and make improvements that we could not carry with us. Neither option was acceptable. So we decided to shut down our business and go out to our Berry Hill property.

"It was a daring move because a complete new studio had not been attempted in Nashville for about five years. New studios had been opened in converted houses and other buildings, but no studio structures had been built from scratch. But we decided that, despite the economic situation, we'd just jump in with both feet and hope for the best."

"We really had no choice," adds Shipley. "We had looked at some old houses and considered ways we could make a studio by tearing out some walls, and adding on here and there. But there's really no way around it. You can sink thousands of dollars into a house, but when you're done you still have a studio in a house. It's a compromise."

Such were the beginnings. They had a pie-shaped lot — 60-foot front, 70-foot back, 170 feet deep — with an existing two-bedroom house covering about 1,000 square feet. The partners also had some rough drawings of a general studio floor plan dating from the pre-Richey House days. And they had the equipment from their old studio, most of which would be leased to Bullet during

STUDIO DESIGN AND CONSTRUCTION IN THE EIGHTIES

Impressions from Treasure Isle designer Richard Lee

Currently vice president and general manager of Criteria Studios in Miami, Richard Lee began his wide-ranging career in recording as an assistant engineer at Motown studios in his native Detroit, a job he held on a part-time basis while still in high school. After completing the Electrical Engineering program at Michigan State, Lee spent several years as a freelance recording and sound reinforcement engineer, working for Fanfare Sound and Metro Audio, among others. In 1977 he joined the staff at Criteria as assistant chief engineer, and in

1978 he was appointed general manager of Compass Point Studio, in the Bahamas. While at Compass Point he designed and supervised the construction of the facility's second studio. During the year 1980-81 he served as manager of consulting services at Valley People, in Nashville, a position which soon led to a full-time job designing Jimmy Swaggert's teleproduction complex in Baton Rouge, Louisiana. Lee assumed his present duties at Criteria earlier this year.

During the course of interviews for the accompanying article on the design and construction of Treasure Isle studios, Lee had some cogent observations to make on topics of current interest.

On the Design and Construction of Compass Point

"Compass Point was a large budget project, and that was fortunate because there were some special problems which led to special design considerations. We had a major airport less than a mile away, so we had a high airborne noise level to deal with. Depending on wind direction, we could be very close to the direct flight path on takeoff, and the traffic was heavy, running the gamut from light planes to 747s. Also, the whole island is one solid piece of bedrock, so we had a lot of structural vibration to deal with.

"The basic idea at Compass Point was to use a building within a building. We had a totally isolated and floating inner studio and control room made from poured concrete and the interim period. What they did not have is what they needed most: money to build the new studio.

One Step at a Time

The construction of Treasure Isle began with less than $50,000 of ready capital on hand. Most of this money was raised through Morgan Vail, Fred's brother, who is a California-based financial planner. He found a group of nine investors looking for investments with good potential for tax write-offs. Vail steered them toward Treasure Isle, though he did *not* tell them they were investing in a recording studio. As far as the investors were concerned, their money was going into real estate and property improvements.

Since they already had this substantial "front money," in addition to the equity in their studio equipment, the two partners assumed they would have little trouble finding bank financing for the balance needed for construction. They were wrong. The banks tended to see them as two rookie entrepreneurs with less than a year's experience in a business (Nashville was rife with rumor), and supposedly headed into a serious slump. What's more, they had few assets that could not easily be removed in a large van in the middle of the night.

After hearing a lot of No's, and facing a variety of closed doors, the partners finally found a sympathetic ear at First Tennessee Bank. According to Vail, "One officer there, Bob Bussey, really took an interest in Dave and I personally, and in our position. As vice-president he had the authority to put the loan through, and we're indebted to him forever — uh, figuratively speaking!"

Even with backing from the California investors and the bank, Vail and Shipley had barely enough money to keep the project underway. Adequate cash flow was a constant problem, as were some frustrating contractor cost overruns. "I'll be frank with you," confesses Vail. "There were times during construction when I was going out to get cash advances on my Master Card just to keep us from grinding to a halt."

Despite limited resources, the two partners realized it would be penny-wise and pound-foolish to skimp on the design phase of the project. Although they both had plenty of practical insight, through observation, into what makes a good-sounding studio, neither had any formal experience in acoustic design. They had mapped out what they wanted in general terms; now they needed somebody who could put it all together and make it sound right.

Acoustic Design Parameters

"We started calling around various places after we lost our lease at Island," Shipley recalls. "We talked to Tom Hidley's people about design, and a couple of others. Then Tom Behrens, a friend of ours at Valley People, said we ought to talk to Richard Lee, who was working for them at the time. We met with him, worked out a financial arrangement, gave him our basic design ideas, and then he went to work. He made a set of blueprints, then updated them as changes had to be made. He had a free hand in the sense that he didn't have to fix any problems stemming from an already existing building.

"We had already determined the basic building size, where we wanted isolation booths, the kind of live sound we wanted, the control room layout, and we also told him we wanted a control room without a compression ceiling. From

Treasure Isle partners Fred Vail (left) and Dave Shipley in the facility's Trident/Studer-equipped control room.

STUDIO DESIGN IN THE EIGHTIES . . . continued —

eight-inch block, with no coupling between the two shells. The floors were completely decoupled using kinetics isolator systems. There were two separate concrete slab ceilings — one for the studio, and one for the control room. We were able to achieve an STC number of approximately 55 using that technique.

"We also added some special support systems when we built the second studio, which the first one didn't have. One was a 150-kilowatt Caterpillar generating system. This was practically a necessity since the whole island is subject to multiple power failures on a semi-regular basis. So this 150-kilowatt, three-phase filtered system ended up in many cases being the primary power source.

"The room treatment, as far as the basic shells are concerned, was pretty much the same at Compass Point and Treasure Isle. At Compass Point we didn't use the offset [non-symmetrical] concrete block walls strutures because we were not able to convince the structural engineers to do it. So we had to start with some rectangles with dimensions that were basically prime number ratios and, with stud-wall construction, we built our own wall offsets and slanted roof structures. That obviously brought the cost way up because of all the labor and materials involved.

"The Compass Point studio was designed primarily for rock and reggae, although the control room is really suited for any type of music production. It was laid out to be a 46-track room, so we always had dual multitracks in mind when plotting the acoustic treatment. The monitoring system was designed to deliver plenty of low-frequency response to accommodate the reggae clients, yet it gives a very true representation of what is actually on tape. We used polycylindrical diffusers along the back wall, all of which were constructed in our own carpentry shop using multiple layers of hand-milled cypress.

"The new studio room at Compass Point is basically just a scaled-down version of the room that was already there. It is very live in the front half, with a vaulted ceiling, and it is heavily trapped in the back, using ceiling and side wall traps. They wanted a very hard sound in the front and a very controlled sound in the back. And for a lot of reggae work, that proved to be the best solution."

On Studios for Digital Recording

"I've been involved in some digital projects over the past few months, and I've seen some aspects of studio design and construction that were moderately important with analog recording suddenly become *critical* with digital. With digital you have almost 30 dB of [additional] signal-to-noise that you don't have with analog, and that opens up a whole new spectrum of things you have to listen for. If your room is masking those problems, you won't hear them until it's too late.

"I see some radical changes ahead in the studio design game. People will have to be so much more noise conscious than they ever have been before. People will be forced to reconsider the way the signal flows around the room. Microphone cables will have to be shortened to the absolute minimum, and all wiring practices will have to be as close to the optimum state of technology as is possible. People will also have to get into sophisticated RF and hum-field shielding in their equipment rooms. Mechanical systems will have to be addressed to find equipment that is much quieter.

"We're going to need a whole new generation of digital studios. I'd say 90% of the studios out there now are unfit for working with digital technology; it's going to be a rude awakening for studio owners.

"With digital technology the tape machine suddenly becomes the strongest link, and the console and the studio systems become the weak links. In addition, you've got all your outboard gear and echo devices which degrade the signal. All of a sudden you've got a great tape machine, but everything else you've got to think about scrapping. It's a tough one to swallow, but it's a reality. It will be good for the vendors out there, provided they can come

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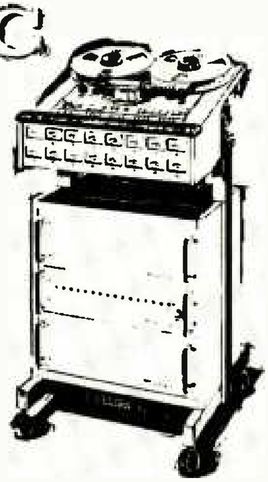

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STUDIO DESIGN IN THE EIGHTIES . . . continued —

up with good new products — and provided the studios can afford to do all of this re-equipping.

On His "Dream Studio"

"Location would be one of the most important factors. You can only spend so much time in the studio, so the surroundings once you get out of the studio are very important. I would want a very large room, something on the order of 50-by-70 or 30-by-60, with very high ceilings. For the control room I would stick with the kind of design I'm presently using: the LEDE-concept with floors about 20-by-22, and with a ceiling of perhaps 16 feet. I think that would be optimum for most people."

On London Studios

"There's a whole different philosophy in England. In London there is no space left, so the majority of studios are renovations of existing buildings. But rather than going into an office structure with, at best, 12-foot ceilings, most of the successful English studios go into places like old churches and movie theatres, where there is already a large volume of space. The English are more conscious, as a rule, of the 'room sound' being part of the overall sound of the record. The American engineer is more geared toward effects and technique rather than the true art of recording, which is mike selection and placement.

"In England there's not as much attention to specific schools of acoustic design. There are only one or two people in London who specialize in that kind of work, and the majority of studios don't even use them. They tend to draw on the ideas of their own engineers, rather than adopt, say, a standard design. That can be good or it can be bad, depending on the background of the people they have."

On Caribbean Studios

"The Caribbean is a unique recording market. For the most part, it's 'mom-and-pop' operations. They are outgrowths of musicians and local entrepeneurs. Most of them are technically as [bad] as you can get, but they have some special people who know how to work in that environment and really create the magic. Kingston is like Nashville, in the sense that it's not a huge city and there is practically a studio on every corner. Because the dream of all the groups in Jamaica is to own their own studio. The rooms are usually very small, with minimum acoustic treatments." ■■■

there he fell right into it. There was good communication from the start. He made suggestions about the best way to go about it, and then calculated the specific dimensions for optimum acoustic results, such as precise wall angles, and exact ceiling heights. For example, we might say we wanted a 25-foot ceiling, and he'd come back and say it would work better if it were only 23 or 24."

Years before, Vail and Shipley had done some research to determine the ideal room size. Drawing on their own observations, and by checking magazines and directories, they discovered that a room roughly 30 feet by 40 feet seemed to be the most common among successful pop/rock studios.

"We definitely emphasized a pop direction in our planning," Shipley offers. "Not that we were excluding country, but you have to have a certain type of studio to do rock 'n' roll. There are more and more people in town now who are looking for that kind of sound, and you can only get it with a *large*, high-ceilinged room."

The decision to go with a big room dramatically increased the overall size of the facility. According to Vail, the concept didn't change that much from the pre-eviction days; it just grew in size. "We originally thought of adding on to the house with a 700 to 800 square-foot studio and control room. As it turned out, we added on about 4,100 square feet, so with the house it's now about 5,100 square feet."

Before construction began, the two partners decided to open the new facility with a new name: Treasure Isle. It's not that they didn't like the old name, Island Recorders. But it seems that Island Records didn't like the old name,

CONTROL ROOM EQUIPMENT

Console: Trident 32-in/24-out Series 80, with separate 24-channel monitor section assignable to input for mixing.
Tape Machines: Studer A80VU 24-track; Studer A80 RC half-inch two-track; Studer B67 quarter-inch two-track; Nakamichi cassette deck.
Monitor Speakers: Custom two-way with dual JBL Model 2235 15-inch woofers, and TAD drivers with Northwest horns.
Monitor Amplifiers: UREI 6500 for low-frequency drivers; and Revox A740 for HF horns.
Close Reference Monitors: JBL 4311; Tannoy; Electro-Voice Sentry 100; Little Reds; Big Reds with Mastering Lab crossovers; Auratones.
Reverberation: AKG BX20 spring reverb; Sony DRE2000 digital reverb; 2,000-square-foot live chamber; Sound Technologies Ecoplate reverb.
Outboard Effects: two Lexicon Prime Times; Audio + Design Scamp rack with 13 effects modules; ADR Vocal Stressor; four dbx limiters; three UREI limiters; Eventide Harmonizer; Eventide Instant Flanger; Orban De-Esser; 24 tracks of Dolby and dbx noise reduction.
Microphones: Wide selection of Neumann, AKG, Shure, E-V, Crown PZM, Sony, RCA, Bruel and Kjaer models.

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The Oak Ridge Boys

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and had begun legal action to express their corporate displeasure. Although confident they were in the right, Vail and Shipley checked the legal defense fund, counted the pennies, and decided the name "Treasure Isle" sounded just fine.

Studio interior with view into the control room, and illustrating the "nautical air" of the acoustic treatment and Fred Vail's detail finish.

Construction and Fabrication

Groundbreaking for the new studio took place in October, 1981. The site was cleared and levelled, crushed rock sub-base put down, and PVC conduit laid for all electrical and audio cabelling. Separate concrete slabs then were poured for the drum booth, studio floor, control room, and lounge/bathroom area. All four slabs were isolated from one another for minimum transmission of sound vibrations. The control room floor was raised for further decoupling, but double flooring was not deemed necessary for the studio, since the building is located in a quiet neighborhood distant from major traffic arteries.

The outer concrete block shell is perhaps the most unique — and most striking — aspect of Treasure Isle's design. Instead of building a rectangular outer structure with acoustic treatment on the inside, Lee recommended building the entire outer structure with multi-angled, non-parallel walls. All walls were constructed from 12-inch block to meet Lee's acoustic specifications, and were then filled to the top with a total of 120 tons of sand.

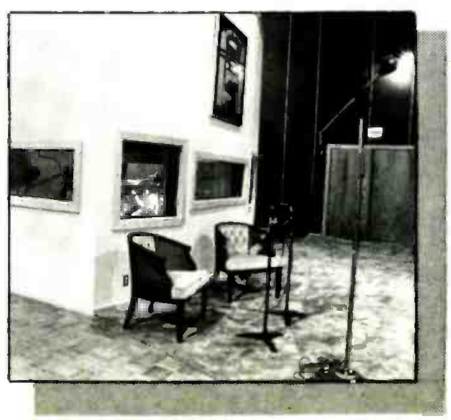
"It was really amazing when we got the masons out here to do it," says Vail. "They were old, traditional guys — one in his fifties, the other in his sixties. They could not believe we were actually building walls at these weird angles, and then filling them with sand as we went up!"

Since the reflective angles already were set by the block walls, the interior was finished by simply attaching 2×4 studs over the block, adding a layer of plywood, and then putting on the finish layer of pine. The soft areas were finished with 2×4 studs, and four inches of #703 fiberglass covered with fabric.

The studio ceiling is suspended from the wood trusses that support the outer roof. Air conditioning ducts are placed below the trusses, and then an outer ceiling with a layer of soundboard (dense fiber panels) and sheetrock seals off the duct area. Next comes a three-foot space with hanging fiberglass batts at different angles, with fabric stretched over the bottom. The ceiling is acoustically dead except for an area of about 17- by 30-foot (the string area), which has a hard ceiling of plywood and finished pine. Floors are parquet and carpet throughout, with large area rugs available for selective deadening.

The extra-large drum booth (12- by 14-foot, with a 13-foot ceiling) is built with double thickness layers of sheetrock inside and out. Building Nashville's largest drum booth was primarily Shipley's idea. "We wanted the drum booth

Central 14- by 12-foot drum booth



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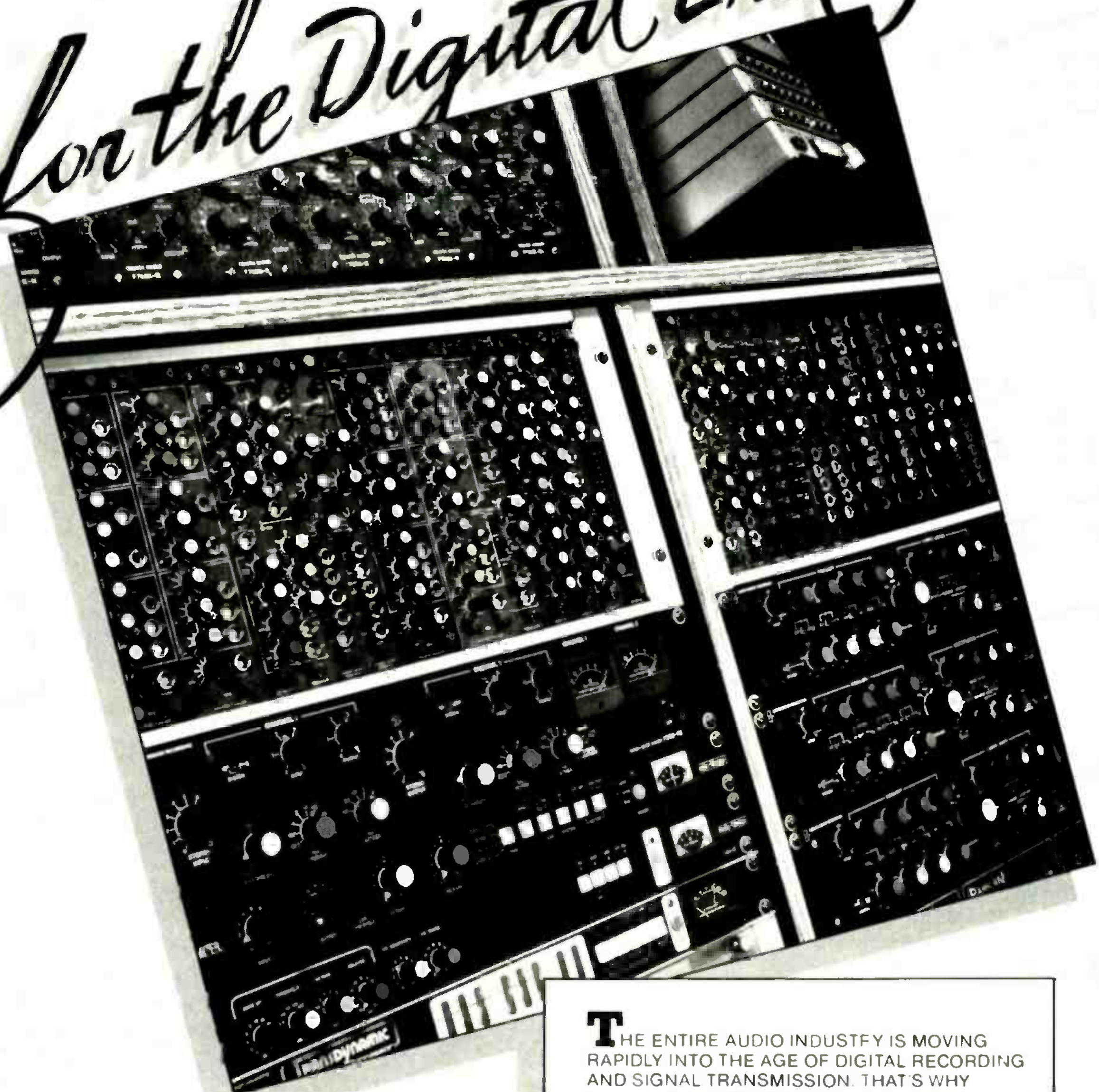

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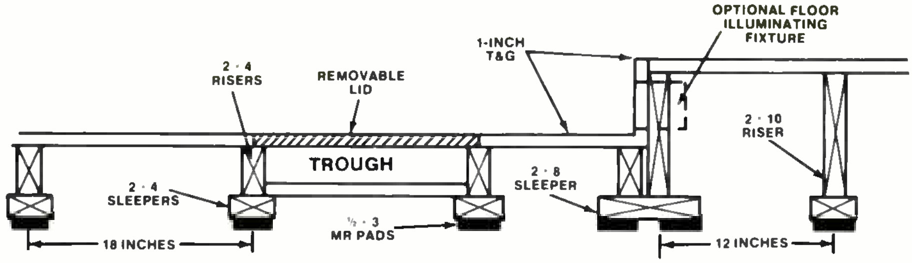
to be large enough to let the drums 'breathe,' " he states. "We wanted to have the effect of having the drums out in the room, yet still retain complete isolation."

The studio, control room, and lounge/bathroom structures are completely decoupled, with approximately one-inch of machine rubber separating the concrete block walls. "It's a bit scary," Vail admits, "but we're still waiting for the first Berry Hill earthquake!"



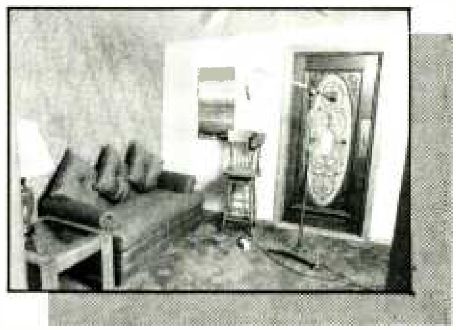
CONTROL ROOM RAISED PLATFORM AND CABLE TROUGH DETAIL

Control Room Design

Design of the control room represents a happy marriage of Shipley's monitoring preferences, and Lee's firm belief in the advantages of the LEDE™ approach. "We decided that we did not want a compression ceiling," says Shipley. "We wanted to have direct speaker sound, without it reflecting off hard surfaces. Naturally that required deadening the area in front of the console. But we didn't want to be mixing in an anechoic chamber; we wanted some 'life' in the room. We explained that to Richard, and he said what we want is a Live-End/Dead-End room. I hadn't read much about the theory behind it at the time; it just seemed to evolve naturally from the kind of sound we were looking for."

Treasure Isle's control room, measuring 22 feet by 24 feet, is very spacious, allowing free movement around the console area. The 13-foot ceiling gives an effect of airy openness, which is in marked contrast to the closed-in feel of many mixing rooms.

"The high ceiling is part of my standard design approach," says Lee. "In LEDE design there are some fixed rules for length and width in relation to the console area, but ceiling height doesn't really enter into it. I like to go for the 12-to 14-foot unfinished ceiling. It gives the low frequencies enough room to expand. You don't have room modes overlapping, so you get true low-frequency propagation."



Vocal and overdub booth

The soffit on the dead end is constructed from 2×6 framing, sheet rock, Soundboard, and other dense materials. For maximum rigidity, all framing is done with bolts, and fastened with 0.22-inch cartridge-driven fasteners. The frame is covered with two layers of ⅝-inch sheetrock, with three inches of Sonex acoustic foam glued to the top. According to Lee, the wedged surface of Sonex has four times the surface area of flat foam, and thus gives the material an extremely high sound absorption coefficient.

The "live," or rear, end of the room also was rigidly framed, with all inward facing surfaces carefully placed to conform with LEDE design rules. The live surfaces, some of which double as cabinet doors, are covered with sheetrock, a layer of half-inch plywood, and

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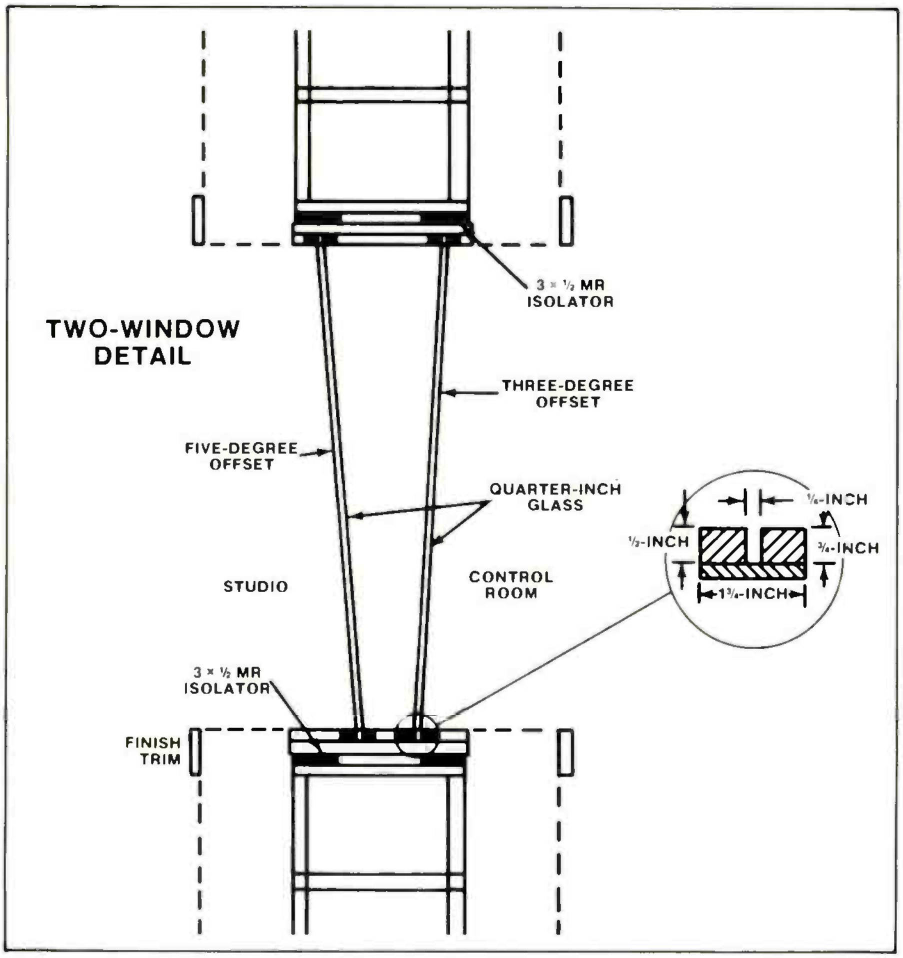

finished pine.

Because all reflections from the rear must be taken into account in LEDE design, Treasure Isle's multitrack (a "wide-body" Studer A80VU with all 24 tracks of electonics in the penthouse) received special consideration. According to Lee, because the Studer is such a "densely-made" machine [read: "solid" — *Ed.*], it acts as a good acoustic reflector. Consequently, placement of the recorder became an integral part of the room's acoustic design.

Monitoring System

Designing the room monitors for Treasure Isle required some preliminary detective work by Shipley. "We liked the Westlake-style double woofer system," he says, "but we didn't particularly care for the wooden horn. I had heard some speakers over at Sound Stage Studio, in the back room, that George Augspurger had designed. I thought they sounded great, but I didn't recognize the horns in them. I spent a month and about $100 in phone calls trying to find out who made these horns. I ended up calling Augspurger, though I felt funny about it since we couldn't pay him to do our speakers. But he was real nice about it. It turns out they were made by Northwest Sound."

The two-way speaker enclosures were designed by Lee in conjunction with Shipley. The woofers are JBL Model 2235 low-frequency drivers, and the horns are mounted on TAD drivers. The 13-cubic-foot enclosures are isolated from the soffit by spring and neoprene mechanical isolators made by Mason Industries. This decoupling prevents very low frequency signals from travelling through the structure ("early-early sound"), and arriving before the signal coming through the airspace.

After the enclosures were mounted in the soffits, the high-frequency drivers were physically aligned with relation to phase and time by mounting them on a sled arrangement, so they could be moved back and forth. Program material was swept through in the octave surrounding the crossover point (800 Hz) while the horns were moved, which caused "burbling" when the woofer and the horn were working the same frequency out of phase. The drivers then were fixed at the point of minimum burbling.

Another interesting offering at Treasure Isle is the planned "wet" reverb chamber. As a dedicated follower of recent English rock albums (the Alan Parsons Project, in particular), Shipley was intrigued by the reverberation sound on many of these records, a sound which seemed distinctly different from all reverb devices he'd heard. Inquiring about, he heard rumors that one of the studios in question had underground live chambers which, through seepage, had partially filled with water. Although unable to confirm this theory, he figured it would be worth a try, so a "wet" reverb chamber became part of the Treasure Isle design.

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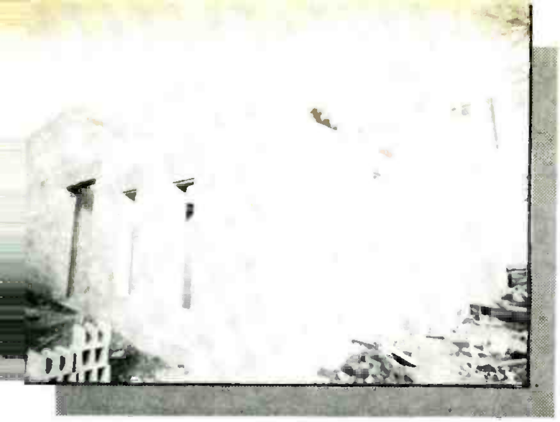


Building shell with multi-angled walls.

The partners originally had hoped to put the chamber underground, but the solid bedrock a few inches below the surface made this option prohibitively expensive. Instead, the 2,000-cubic-foot chamber ended up sitting astride a hallway, with special structural supports underneath to allow the addition of up to six inches of water. (As of this writing the chamber had yet to be "baptized," so the outcome of this experiment is still pending.)

Treasure Isle's electrical system consists of both standard and tech power systems. The tech power is a 100-amp feed from the main power line, which passes through an isolation transformer. All recording equipment is on this circuit, and tech power may also be accessed in the studio through outlets designated by stainless steel covers.

To prevent noise bleed-through, three separate air-conditioning units are used: one for the studio; one for the control room; and one for the lounge area.

From the outset, the entire Treasure Isle complex was conceived as a total recording environment. If you want to set up drums by the Jacuzzi, and put your guitar amp in the tape storage room, it's no problem. More than 65 mike inputs are spread around the building; 32 run direct to the console, the remaining inputs being patchable. Nine headphone distribution boxes, each with three outputs for the two separate cue systems, also are spread throughout the studio, control room, lounge, and whirlpool/sauna room. The PVC conduit runs for electrical, cue, and microphones are completely separate and spaced as widely as possible. Wiring diagrams for Treasure Isle were done by Tom Behrens of Valley People, in consultation with Richard Lee. Installation was done by Shipley, Treasure Isle's former staff engineer Dave Hieronymous, Tom Behrens, Jeff West, Richard Achor, and Deborah Bradley.

First Sessions

After a few months of fine-tuning and finishing touches — and after signing a new 20-year mortagage to retire their construction debts — Shipley and Vail now consider their mission accomplished. The studio already is drawing a growing list of Nashville clients, and out-of-town producers and artists from Los Angeles, New York, Florida, and Washington have completed projects at Treasure Isle. It was a risky venture in the beginning, but the duo felt sure they would pull it off — just as long as they were not forced to compromise.

"I always felt there was room for another *great* studio in Nashville," says Shipley. "If I'd thought we'd have to compete at the same level as most studios here — or around the country, for that matter — I might have given up somewhere along the line. But I knew we could come up with something better. There's really no big secret to it. It's just a matter of concentration, of paying attention to what people want, and really caring about doing a better job."

In this regard, it's often the little touches that separate the extraordinary studios from the run-of-the-mill operations. Consider, for example, the small brass plates adorning the doorways at Treasure Isle. Located by Fred Vail in a ship's store in New Orleans, the door plates were scavenged from an old ship. The one over the bar states succinctly, "Cert. For Liquor Locker." On the door of the drum booth another proclaims, with multiple connotations, "Head Room." Although no propellers or machinery are stored in the studio sauna/whirlpool room, the door plate reads quizzically, "Screw Room." And finally, considering all Shipley and Vail have endured in the course of construction, the brass inscription over the control room door is most appropriate: "Cert. For Mental Ward." ■■■

STUDIO UPDATE

FACILITIES
EQUIPMENT
PEOPLE

Northeast:

☐ **THE BROCCOLI RABE RECORDING STUDIO COMPLEX** (North Haledon, New Jersey) has relocated its recording studios, record company, and publishing company under one roof at its newly opened 40,000 square foot facility located nearby the Caldwell Airport. The new complex offers office space, rehearsal rooms, recording studios, and a video studio. The music control room features an Otari MTR-90 16/24-track with remote and autolocator, and Tascam two-track mastering machines, fed through a Tascam M-15 24/24 console. Outboards include 28 channels of dbx noise reduction, a complete line of dbx compressor/limiters, MICMIX Master Room XL-305 reverb, a Delta Lab DL-2 Acousticomputer, Roland Space Echo, and a Valley People effects rack. Mikes are by Shure, AKG, Neumann, and Sennheiser. The instrument list boasts Ludwig and Slingerland drum kits, two Yamaha grand pianos, Yamaha CP-30 electric piano, Hammond M-3 organ with Leslie, and wide selection of percussion instruments, including Roto-toms, according to studio president **Brian Drago**. *184 Ballentine Drive, North Haledon, NJ 07508. (201) 427-8316.*

☐ **BAYSIDE SOUND** (New York City) has completed installation of an MCI/Sony JH-636 automated console, and a JH-24 24-track machine. The installation, supplied by Audiotechniques, also included an Eventide H910 Harmonizer, UREI LN1176 limiters, Valley People Dyna-Mite gates, and an Otari MX-5050 two-track, according to studio manager **David Eng**. Currently on order is an MCI/Sony JH-110 half-inch two-track mastering machine. *200-70 39th Avenue, New York, NY 11361. (212) 225-4292.*

☐ **WESTRAX RECORDING** (New York City) in a recent upgrade from 8- to 16-track, now boasts a Sound Workshop Series 30 console interfaced with a Tascam 85-16B 16-track equipped with dbx noise reduction. Outboards include a Lexicon PCM-42 digital delay unit, and an Orban parametric EQ. In addition, a new isolation booth is nearing completion. **Peter Link** is the owner at Westrax, while **Bob Lowe** serves as managing director, and **Jesse Plumely** head engineer. *Manhattan Plaza, Basement Level, 484 West 43rd Street, New York, NY 10036. (212) 947-0533.*

☐ **ALPHA INTERNATIONAL** (Philadelphia, Pennsylvania) has fitted Valley People ECG202 Class A VCAs in both of the facility's Harrison 4032 and 2824 consoles, and also restructured the patch-point signal levels. New equipment includes an E-mu Systems Emulator, Dyno My Piano and Tri Stereo Chorus Rack, Oberheim DMX digital drum machine, and outboards from Lexicon, Valley People, Aphex, AKG, Carver and Tascam. *2001 West Moyamensing Avenue, Philadelphia, PA 19145. (215) 271-7333.*

Southeast:

☐ **OMEGA RECORDING** (Kensington, Maryland), has acquired a second 24-track music studio located in downtown Washington, D.C. Formerly know as "Room 10," the new facility is still serving its regular clientele, while also handling the overflow business from Omega's suburban studios. According to owner **Bob Yesbek**, the new studio features MCI console and tape machines, and UREI monitors. Omega plans to centralize both of its operations into one large three-studio complex in the near future. *10518 Connecticut Avenue, Kensington, MD 20795. (301) 946-4686.*

Southcentral:

☐ **SIERRA RECORDING** (Fort Worth, Texas) has added a Sound Technologies Ecoplate III reverb unit, two UREI LA-4 limiters, an Orban 6228 stereo parametric equalizer, and an Orban Model 245E Stereo Synthesizer. The 24-track studio, opened in October 1982, has gained a reputation for Spanish language recording with clients from the US, Mexico, and South America. *669 Seminary South, Fort Worth, TX 76115. (817) 921-3881.*

☐ **BULLET RECORDING** (Nashville) currently is gearing its operation for a transition from an audio and video hardware rental house, to

3M AND MITSUBISHI DIGITAL DELIVERIES

☐ **THE BENNETT HOUSE** (Franklin, Tennessee) has installed a second 32-track 3M Digital Mastering System, which joins the first digital 32-track delivered three months ago. The facility also boasts a pair of 3M digital four-tracks for mixdown, and a full 3M Digital Editing System. The two-room complex, which features a 28/24 Trident A-Range console in one room, and a 32/24 Trident Series 80 in the other, recently upgraded its control-room monitors with the installation of John Meyer Model 833 systems that utilize a MS15 15-inch woofer on the bottom-end, and an MS1404 driver for the upper frequencies. Commenting on the recent acquisition of a second digital multitrack, studio president **Norbert Putnam** says, "We went with 3M because the company recently has made a tremendous price reduction, and the digital system is now very cost-effective. In the past we used to lock up for 46-track sessions, but the cost of a pair of 24-tracks, plus the SMPTE synchronizer, is pretty high these days. But the 3M 32-track has a good price advantage. Now that there are around 100 3M systems in use around the world, compatibility between studios is less of a problem —we have clients that want to take their tapes to London, New York, and so on. It's the best sounding recorder I've ever heard. Since we went digital, the demand on studio time has tripled — we've never had to turn away so much business before!" *134 4th Avenue North, Franklin, TN 37064. (615) 790-8696.*

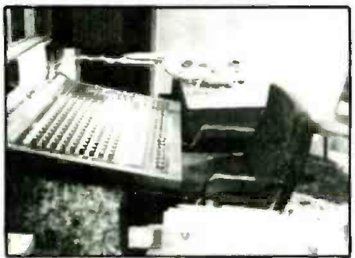
☐ **LION SHARE STUDIOS** (Los Angeles) has installed a Mitsubishi digital recording system, comprising a Model X-800 32-channel recorder, and X-80 portable and X80A console two-channel mastering machines; an XE-1 electronic editing system is scheduled for delivery in the near future. The total contract was supplied by **Digital Entertainment Corporation**, and is said to be the first Mitsubishi digital audio recording package to be installed at a major studio. The choice of Mitsubishi digital by Lion Share is due, according to **Terry Williams**, Lion Share studio director, to the extraordinary sound quality, together with the ability to record 32 tracks of digital audio, and the electronic editing system's ability to work with both multitrack and two-track formats. Lion Share now has the capability of maintaining first-generation audio from multitrack through the complete studio chain, including disk and cassette mastering all the way to the consumer via Compact Disc. In the facility's Studio A, an US Festival '83 project recently was completed where 24-track audio, using a Neve NECAM automation system and an Ampex VPR2 one-inch VTR, was mixed to picture. Overdubs also were added. At the same time, Studio B is equipped to mix, overdub and sweeten to picture by linking the studio to the video facility. *8255 Beverly Blvd., Los Angeles 90048. (213) 658-5990.*

LION SHARE — Mitsubishi X-800 digital mulitrack

a creative production company for both film and music. The new company, *Bullet Creative Group, Inc.*, will include on the staff Dallas-based film director **Bill Buhcanan**, **Merissa Ide** as production manager, and **Lora L. Brown** as assistant to company president **Randy Holland**. *49 Music Square W, Nashville, TN 37203. (615) 327-4621.*

Midwest:

☐ **LANSING SOUND STUDIOS** (Lansing, Michigan) has opened its brand new Studio B, a "state-of-the-art" eight-track recording facility located in the new WSFL-TV building. The facility is designed for commercial and small scale music production. **Bob Baldori** is the owner/operator of Lansing Sound, with Studio A located in Okemos. *616 West St. Joseph, Lansing, MI.*


STUDIO B — new Tascam 38 eight-track

☐ **STAIRWAY RECORDING STUDIO** (North Kansas City, Missouri) is an eight-track facility featuring a TEAC 80-8 multitrack and a 25-2 mastering machine, both equipped with dbx noise reduction, interfaced with a Tascam Model 5A-5EX console. Monitoring is handled by JBL 4311s and Auratones powered by a Yamaha P2200 amp. Outboard gear included a Lexicon Prime Time Model 93 digital delay, Orban 111B Reverb, Bi-Amp Quad Limiter, Nikko 10-band graphic EQ, and Ross noise gates. Mikes are by Sennheiser, Shure, Sony, and Electro-Voice. The instrument list boasts a Stage 73 Rhodes electric piano, Roland Juno 60 synthesizer, and Rodgers drum set. Record pressing services also are available, according to studio manager **Mark Hubble**. *204 East 18th Avenue, North Kansas, MO 64116. (816) 421-1180.*

☐ **STUDIO B** (Rockford, Illinois) has expanded to eight-track operation with the addition of a new Tascam 38 multitrack. The Allen & Heath 16/4/2 mixer also has been recently custom modified to handle eight-track sessions. The studio features Bi-Amp compressors and reverbs, an MXR Flanger/Doubler, Shure mikes, and a Real-Time cassette duplication system. The studio is owned by producer/engineer **Michael Castronovo**, and his wife **Deborah**. *2215 Wilmette Drive, Rockford, IL 61108. (815) 398-4477.*

☐ **QCA RECORDING STUDIO** (Cincinnati, Ohio) has added a Studer A80-VU 24-track, an Advanced Music Systems RMX-16 digital reverberation system, and four Aphex CX-1 compressor/expanders. *2832 Spring Grove Avenue, Cincinnati, OH 45225. (513) 681-8400.*


AUDIO VILLAGE — AHB Syncon desk

☐ **AUDIO VILLAGE** (Bloomington, Indiana) recently updated with the installation of an Allen & Heath Syncon discrete 28×24 console, equipped with quasi-parametric EQ and 16 submix channels. Also purchased: Neumann SM-69, Neumann SM-2 and AKG C-24 stereo vacuum tube microphones. In addition, a van has been purchased for location recording, and a computerized labeling system for cassettes, according to **Wayne Gunn**, studio manager. *1000 West 17th Street, Bloomington, IN 47402. (812) 332-7475.*

☐ **SOLID SOUND RECORDING STUDIO** (Hoffman Estates, Illinois) has re-designed its control room, and added a new monitoring system featuring UREI 813-B Time-Aligned monitors, and a newly designed crossover network. Other new gear includes a Delta-Lab DL-4 digital delay, Orban De-essers, UREI Digital Metronome, and a package of Neumann, Sennheiser, and AKG microphones. *2400 West Hassell Road, Suite 430, Hoffman Estates, IL 60195. (312) 882-7446.*

☐ **RAINBOW RECORDING** (Omaha, Nebraska) has upgraded its facility with a newly remodeled control room and 24-track capability featuring the Otari MTR-90 Series II. Operations center around a Stevenson Interface 30-channel console. Also now available is a Lexicon PCM-42 DDL with extended memory option. Instruments include a Sequential Circuits Prophet 5 with sequencer, Linn Drum Machine, and a Baldwin seven-foot concert grand piano. **Rick Schwartz** is the studio manager. *2322 South 64th Avenue, Omaha, NE 68106.*

☐ **AUDIOGRAPH PRODUCTIONS** (Okemos, Michigan) has opened a newly upgraded facility, which features a Studer A80 MkIII 24-track, Studer A810 two-track, and a Studer A710 cassette deck. Mixing is handled by a Neotek Series II 32/24 console complemented by an outboard rack offering a full line of compressors, EQ units, delay systems, and Valley People Gain Brain compressors. The new facility was designed by chief engineer **Glenn Brown**, and features variable acoustics (0.7- to 0.45-second RT60 broadband) with louvered traps. UREI 813-A Time-Aligned units handle the monitoring. **Doug Monson** is studio president. *2810 Bennett Road, Okemos, MI 48864. (517) 332-3272.*

Mountain:

☐ **COLORADO SOUND RECORDING** (Westminister, Colorado) has opened a new facility featuring a Trident mixing console feeding an Ampex 24-track, with monitoring handled by JBL 4311s and UREI 813-As. The 450-square-foot control room overlooks a 550-square-foot live recording area, and a 600-square-foot rhythm room with a large drum booth. The studio was designed by **Milam Audio**, and features an organ and a grand piano on its instrument list. *3100 West 71st Avenue, Westminster, CO 80030.*

Southern California:

☐ **ALPHA STUDIOS** (North Hollywood) has expanded its facilities with the addition of a BTX Softtouch SMPTE synchronization system. The new gear was demonstrated on John Cassavetes' new film *Love Streams*, music and vocals being pre-recorded in the studio for actors and actresses to lip-sync to during filming. SMPTE was utilized from 24-track master to Nagra ¼-inch copy, through the transfer to three-stripe 35mm mag stock for film editing. Editing will be completed on ¾-inch video at Alpha. A new 360 Systems Digital Synthesizer, capable of digitally storing pre-recorded acoustic instruments, including full bandwidth and attack, was utilized furing scoring of the film. *North Hollywood, CA. (213) 760-2825.*

Northern California:

☐ **HYDE STREET STUDIOS** (San Francisco) is now set up for 16-track as well as 24-track mixdowns with the acquisition of a 16-track playback head for the facility's Otari MTR-90, which also can be locked up with the studio's MCI multitrack to handle 30-track mixing. *245 Hyde Street, San Francisco, CA 94102. (415) 441-8934.*

☐ **TRES VIRGOS STUDIOS** (San Rafael) has added another Ampex ATR-100 mastering machine. According to studio manager **Christa Corvo**, the deck is equipped with Strategic Sound's new transformerless input/output cards. New units in the mike line-up include two Crown PZM 315S, RCA 77DX, AKG D20 (Serial #10), AKG D45, and two ST&C 4038s. *1925 Francisco Boulevard, San Rafael, CA 94901. (415) 456-7666.*

Australia:

☐ **STUDIOS 301** (Sydney, New South Wales) has purchased a Sony PCM-1610 digital audio processor, dedicated for use in audiophile cassette and disk mastering. An additional upgrading to one-inch analog mastering tape for cassette duplication is said to provide greater phase stability and dynamic range. An analog improvement includes the availability of half-inch two-track Studer A80 tape machines for disk

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cutting and cassette mastering. Digital recording is also available at Studios 301. *301 Castlereagh Street, Sydney, NSW, Australia 2000. Telephone: 20912.*


HITOKUCHI-ZAKA — multitrack studio

Japan:

☐ **HITOKUCHI-ZAKA STUDIO** (Tokyo) has opened its new Kawaguchi-ko Studio about 60 miles outside the city, near Mount Fuji and Lake Kawaguchi. The facility is equipped with a customized API 2824 console interfaced with two Studer A800 24- and 16-tracks, and two Studer A80 two-track decks for mastering. Monitoring is handled by Super Red double-woofer speakers powered by McIntosh amplifiers. Outboard gear is by Dolby, dbx, UREI, and ADR. Orban compressor/limiters and EMT-140 and EMT-240 reverbs also are to be found in the rack. Effects devices include an Eventide Harmonizer, Lexicon Prime Time Model 93, URSA MAJOR SST-282 Space Station, and MXR phaser. A Fender Rhodes, and Yamaha acoustic and electric pianos also are offered. Mikes are by Neumann, Schoeps, AKG, Sennheiser, Electro-Voice, and Sony. **Yasuji Ono** is the facility's director of engineering. *4-3-31 Kudan-Kita, Chiyoda-Ku, Tokyo 102, Japan. (03) 263-1097.*

Great Britain:

☐ **CTS STUDIOS** (Wembley, England) has completed the first stage of its digital upgrading with the acquisition of the Sony PCM-3324 digital 24-track, and Sony PCM-1610 digital audio processor. The second phase of the re-equipping will take place in early 1984, when a Neve DPS digital console is installed in CTS Studio #1. *The Music Centre, Engineers Way, Wembley, Middlesex, England, HA9 ODR. (01) 903-4611.*


ABBEY ROAD — Townshend & Colin Sanders

☐ **ABBEY ROAD STUDIO 2** (London) has opened following a summer-long renovation of its control room, including the installation of an SSL 4000E Series Master Studio system, equipped with 48 channels of SSL Total Recall input/output modules, to provide full reset capability on up to 96 inputs during mixing operations. According to **Ken Townshend**, Abbey Road's GM, the contract for the new desk was awarded to SSL after a poll of the artists and producers who have used the facility in the past, as well as those acts who are signed to EMI's various labels. During renovation, Studio 2 was open to the public for the first time, visitors being treated to unique memoribilia, film footage and unreleased material by the Beatles, who made the studio their own during most of the Sixties. While the control room has ben completely reworked, the acoustics of the studio itself have been left alone, since it was felt that no improvements could be made. A second SSL System already is on order for Studio 1, the facility's orchestral scoring stage. This console will also be equipped with Total Recall computers, allowing complete freedom and continuity in moving projects between the two rooms. *3 Abbey Road, London NE8 9AY, England. (01) 286-1161.*

— AUDIO/VIDEO UPDATE —

Eastern Activity:

☐ **DU ART FILM LABS** (New York City) has installed a new Harrison TV-3 audio mixing console, which will form the center of a recent re-recording room upgrading that includes new mag dubbers, a four-track 35mm recorder, JBL 4430 Bi-Radial monitors, various outboard equipment, a new projection screen, and a new footage counter. **Martin Audio**, Harrison's New York representative, supplied and installed the new desk, including a special monitor and remote-control designed by Martin engineers. According to Du Art chief engineer, **Dominick Tavella,** "we bought the Harrison because it offers more function and flexibility than any other console we considered, and does so at a very reasonable price." *245 West 55th Street, New York, NY 10019.*


DU ART FILM — Harrison TV-3 console

☐ **ARTISAN RECORDERS** (Pompano Beach, Florida) provided a 40-channel live mix of this year's **Reggae Sunsplash** in Montego Bay, Jamaica, presented by Synergy Productions, Ltd. Among the acts appearing were Musical Youth, Third World, Rita Marley, Black Uhuru, and Steel Pulse. **Peter J. Yianilos** engineered with the assistance of the Artisan crew: Larry Janus, Kevin Ryan, Stan Johnson, Vince Oliveri, Rey Monzon, and Mike Drozd. **Trillion,** the London-based production company, handled the filming of the event. *1421 South West 12th Avenue, Pompano Beach, FL 33060. (305) 786-0660.*

☐ **NATIONAL VIDEO CENTER** (New York City) provided audio mixing for Johnston Film's award-winning *VII International Tchaikovsky Competition*, recently picked up by MasterVision as the premiere offering in its Beta HiFi and VHS stereo videocassette series, which features improved-quality audio playback. *New York, NY.*

Central Activity:

☐ **EYE & EAR TELECORP** (Chicago, Illinois) is the new, fully-equipped film and video tape house aiming at the video-music market. Eye & Ear boasts a new 2,000-square-foot office and recording facility, and a 20-foot mobile video recording unit featuring one-inch VTR for multicamera shoots, a 16-channel audio board and 16-track machine. The mobile unit is housed in a refurbished Superior Motor Coach truck. Other in-house gear include Ikegami broadcast cameras, and a CMX SMPTE timecode editing controller. Recent projects have been a music-video for Disco de Tinga artists **Bohemia**, and live shoots for the **Fabulous Thunderbirds, Larry Coryell,** and **Bauhaus.** *612 North Michigan Avenue, Suite 802, Chicago, IL 60611. (312) 337-5050.*

☐ **OMEGA AUDIO** (Dallas, Texas) recently provided audio services for the video taping of the *Stars For Children* syndicated television special. The show was held at the Reunion Arena, Dallas, where Omega supplied 24-track SMPTE-encoded recording, while **Tele-Image** of Dallas handled the video shoot. Post-production was also provided by Omega at its 24/48-track studio at Dallas Love Field. Appearing at the concert were **The Commadores, Roseanne Cash, Charlene Tilton,** and hosts **The Oak Ridge Boys.** *8036 Aviation Place, Box 71, Dallas, TX 75235. (214) 350-9066.*

Western Activity:

☐ **BONNEVILLE PRODUCTIONS** (Salt Lake City, Utah) handled all the audio post-production for the Osmond Family's July 4th Extravaganza, *The Glory of America*, performed live at Couger Stadium in Provo, Utah, and televised on the Ted Turner Cable Network and Armed Forces Network. Audio post-production was handled at Bonneville's 24-track Studio C, which features an Audio-Kinetics Q.Lock synchronizer, 36-channel Neotek console, and Ampex multitracks. Audio layover and layback operations were done via direct cable connection between Video West and Bonneville, to eliminate tape-generation problems. *130 Social Hall Avenue, Salt Lake City, UT 84111. (801) 237-2600.*

STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

☐ **THE COMPLEX** (Los Angeles) provided sound stage facilities for the filming of Paul Kantner's upcoming music-video, *Planet Earth Rock 'n' Roll Orchestra.* Grace Slick and Kantner and Slick's daughter China also are featured on the piece, directed by **Arthur Ellis** for Limelight Productions. *2323 Corinth Street, West Los Angeles, CA 90064. (213) 477-1938.*

☐ **HARLEQUIN STUDIOS** (Northridge, California) has completed its new video complex featuring a full two-story, 35- by 40-foot main studio, two dressing rooms, and control rooms with Sony half-, ¾-, and one-inch videotape formats. *19347 Londelius Street, Northridge, CA 91324. (213) 993-4778.*

ON THE STUDIO TRAIL

Mel Lambert at Large this month in Chicago's Commercials Production Scene

While many would consider the Big Apple to be this country's advertising and commercials capital, there is little doubt that in terms of multipurpose production facilities at least, Chicago handles a large proportion of the TV and radio jingle recording business. As I discovered during a recent visit to the Windy City, the commercials production market currently is in a very healthy and competitive state, with many studios looking to increase their share of this lucrative market. The facilities I managed to visit while in Chicago included **Audio Mixers, Universal Recording, Streeterville Studios,** and **Chicago Recordings.**

As well as handling audio recording for educational and industrial multimedia productions, Audio Mixer's owner Steve Schwartz tells me that his studio has played host to several scoring sessions for commercials and film soundtracks. Bookings run about 50% for radio and television commercials, the remainder being taken up with audio/visual work. His Rush Street facility comprises a compact studio with drum booth, plus a separate isolation/vocal booth. Although the main room isn't exactly huge, Schwartz says that with careful mike techniques he can easily accommodate 30-piece choirs.

Recording hardware centers around an 18-channel Tangent console linked to a Tascam 80-8 half-inch eight-track with dbx, and several Ampex AG350 mono/stereo mastering decks (also equipped with dbx noise reduction); outboards include an Audio + Design Compex compressor/limiter, a pair of UREI LN1176 limiters, Orban Stereo synthesizer and Sibilance Controller, and an Eventide H910 Harmonizer. Monitoring is handled by Crown-powered JBLs.

Schwartz offers that for commercials dates, eight-track is the *minimum* format he would consider. His track sheet for such sessions looks something like this: narration on track #1; music tracks A and B across tracks 2 and 3 (or 2 thru 5 if both were tracked in stereo), to allow for cross segues between musical phrases; sound effects (wind, traffic noise, etc) on another track; isolated spot effects (door slams, phone rings, and all the rest) on another; and remaining tracks as required. Music tracks are taken from needle-drop libraries, or recorded eight-track in the studio, mixed to mono or stereo, and then laid back onto the eight-track for later addition of pre-recorded effects (from disk libraries or reel-to-reel) and live narration.

. . . continued overleaf —

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

AUDIO MIXERS/Steve Schwartz

Currently, he uses stopwatch timing when mixing audio for video, although at the time of my visit Schwartz was considering the new Tascam Model 58 eight-track for locking up multitrack audio to a videocassette transport via SMPTE timecode interlock for more demanding synchronization sessions.

Regarding processing and recording techniques for radio and TV, Schwartz offers that broadcast audio invariably ends up being heavily compressed by the station. It is essential, he feels, that a certain amount of compression be added during the mix stage, if only to judge how the music/narration balance will be affected by subsequent broad-band compression. Also, to help a voice track cut through a background music bed, he inserts an EQ notch in the latter at the center of the announcer's vocal range, and then further boosts the voice in the final mix.

Across town at Universal Recording, Murray Allen gave me a conducted tour of his diversified facility, which stretches across a couple floors in adjacent buildings on East Walton, and includes several production and recording studios, 16 and 35mm mag and optical transfer rooms, two film theaters, various copy rooms, residential apartments for visiting producers, a woodwork shop, and a tape-duplication plant. A separate building in the downtown area also houses a couple of commercial production rooms. The facility even keeps a drum technician on staff to look after all the studio-owned trap sets.

Allen reports that 70% of studio time is taken up by commercial and jingle production, and 30% on "conventional" record dates. Studio A comprises an extremely large recording area capable of accommodating 40 to 50 string players, and featuring a movable drum booth, working into a spacious control room that houses a Neve 8048/78 console with NECAM disk-based automation, and a pair of MCI JH-24 multitracks. A BTX Shadow SMPTE synchronizer provides interlock between videocassette decks and the 24-tracks, the latter having been modified, Allen says, to operate in chase mode at 80 IPS for faster setups. Studio B — the "SSL Room" — also is equipped to handle 24- or 46-track sessions, and is located beneath a 35mm projection room which, in addition, houses a 3M 32-track Digital Mastering System capable of being patched and operated from either Studio A or B.

Other rooms of interest around the complex include the "Back Room," which is equipped for overdubs, remix and audio layback to two-inch Quad or one-inch C-Format videotape. Production hardware centers around an automated 28-input MCI JH-628 console, Ampex MM-1200 and MCI JH-24 multitracks, BTX SMPTE synchronizer, and UREI 813 monitors. A small room and a couple of isolation booths attached to the control room enables voice-overs and small rhythm sections to be recorded without occupying one of

STREETERVILLE/Jim Dolan, Jr.

CHICAGO RECORDING/Hank Neuberger

the larger studios.

Turning to the subject of digital recording for commercials production, Allen is a strong believer in utilizing the improved sound quality and dynamic range available with the 3M 32-track digital system. Although rhythm tracks might be laid on analog to achieve the familiar kind of sound that a producer is after, the basics then would be transferred to digital for overdubbing. With digital, he offers, you have many more tracks available for basics plus vocal and solo overdubs. In addition, vocals can be laid off onto four-track digital, and then slipped to another part of the multitrack digital tape against SMPTE timecode locations, without any degradation in signal quality. Also, if a client is looking to record, for example, several versions of a particular commercial that share common music tracks, with maybe different vocal treatments and/or lyrics, identical digital copies can be made of the basic tracks without the quality losses associated with analog-to-analog dubbing. Digital also enables tracks to be recorded without compression, and still have up to 40 dB of extra dynamic range, available during remix to add extra EQ for effect.

At Streeterville Studios on East Grand, studio manager Jim Dolan, Jr. showed me over the five-room complex which, he says, offers "compatible room acoustics" to make it easier for a project to move between studios without the sound character of tracks altering from environment to environment — all of which can help speed up a session, he feels, and maintain continuity between recording basic rhythm tracks in one room, and string/horn/vocal overdubs in another. Acoustic design in the various rooms has been towards what Dolan refers to as "controlled ambience" — open drum cages, tall ceilings, and plenty of working space.

"Music 1" features a large recording area with drum and isolation booths, coupled to a control room equipped with an automated 48-input Neve 8108 console with NECAM II, MCI JH-24, UREI 813 monitors, and Sony U-Matic videocassette for sweeting or mixing to picture. "Music 2" is laid out as a self-contained facility separate from traffic in the rest of the complex, and features a 40-input Harrison 4032C Series transformerless console, and a large studio area complete with skylight and isolation booth for a piano or small string section. Studios B and C are smaller four-track rooms designed for radio, TV, and multimedia production, and feature identical custom consoles with interchangeable EQ modules, Studer A80 four-track on half-inch machines, and B67 mastering decks.

The "Remix Suite" is set up for both conventional music mixdown, and/or video sweetening, and houses a 40-input Harrison 4032B console with Auto-Set automation, a pair of MCI JH-24 multitracks, Sony BVU-800 U-Matic, Audio Kinetics Q.Lock 3.10

UNIVERSAL RECORDING/Murray Allen

STUDIO UPDATE
FACILITIES EQUIPMENT PEOPLE

SMPTE synchronizer, Otari MTR-10 four-track mastering deck, and UREI 813 monitors. An associated studio is capable of hosting a 6- or 8-piece string or brass sections, or for use during vocal overdubs.

And regarding the type of mix balance and processing required for radio and television commercials, as opposed to conventional record dates, Dolan says that while there is a definite sense of "mixing for the medium," this depends to a great degree on the "taste of the producer — he or she might be looking for a 'record-style' mix, or wish to emphasize a band arrangement, or vocal treatment," he offers. "But quite often a commercials producer will come up with several different types/styles of mix, which he'll play to an agency, and then come back and make a few vocal or mix changes. In this industry, console automation is a real *must*, because we might go through as many as 20 or so different mixes until the client is happy with the end result. That's also why it makes sense for us to have a dedicated remix room, so that a studio won't be tied up when the producer has to rework a mix.

"Audio will always be important to television commercials," Dolan concludes. "People don't go around humming the picture—it's always the song or the tagline that people remember!"

Alan Kubicka's Chicago Recording ranges across several floors of adjacent buildings on Michigan Avenue. The entire complex comprises a multitude of recording and production studios, plus a 35mm mag transfer room. Starting my tour in Studio A, operations manager Hank Neuberger pointed out that the control room recently had been enlarged by removing the pair of rear-mounted quad speakers, and extending the back wall about six feet. The control room, he recalls, was the last environment designed by Tom Hidley under the Westlake guise (prior to his relocation to Europe, and Eastlake persona). Currently the room, which houses an MCI JH-532 console and JH-24 multitrack, is used about 40% of the time for record dates, and the remainder for scoring, commercials, voice-over, and audio/video sweetening.

Studio B, which sees more record sessions that other rooms the facility, centers around a Neve 8068 with NECAM automation and in-line monitoring, Studer A80 and MCI JH-24 multitracks, BTX Shadow synchronizer (an MCI JH-45 synchronizer being on hand for JH-24-to-JH-24 lockups), and Cadac monitors. (The entire control-room package is scheduled to be relocated to a different building during this Fall; Studio B will be re-equipped at that time with as yet unspecified hardware.)

Studio D features a large string area, complete with tall ceilings, hard-wood floors, and mirrored walls. Two isolation booths and a drum booth also are available, enabling a full orchestra, horn/rhythm and vocal sections to be tracked simultaneously, if necessary. Control-room hardware includes a Cadac 32/32 console, MCI JH-24, UREI 813 monitors, ¾-inch U-Matic videodeck for interlock to picture, and a Sony one-inch videotape transport for audio layback.

A collection of purpose-built production studios — a couple in the main complex, and three in an adjoining building — accommodate a variety of commercials work. MCI JH-110 eight-, four-, and two-tracks handle recording duties, while the consoles include a custom-designed model, a Sphere, a Neotek, and a Trident Trimix.

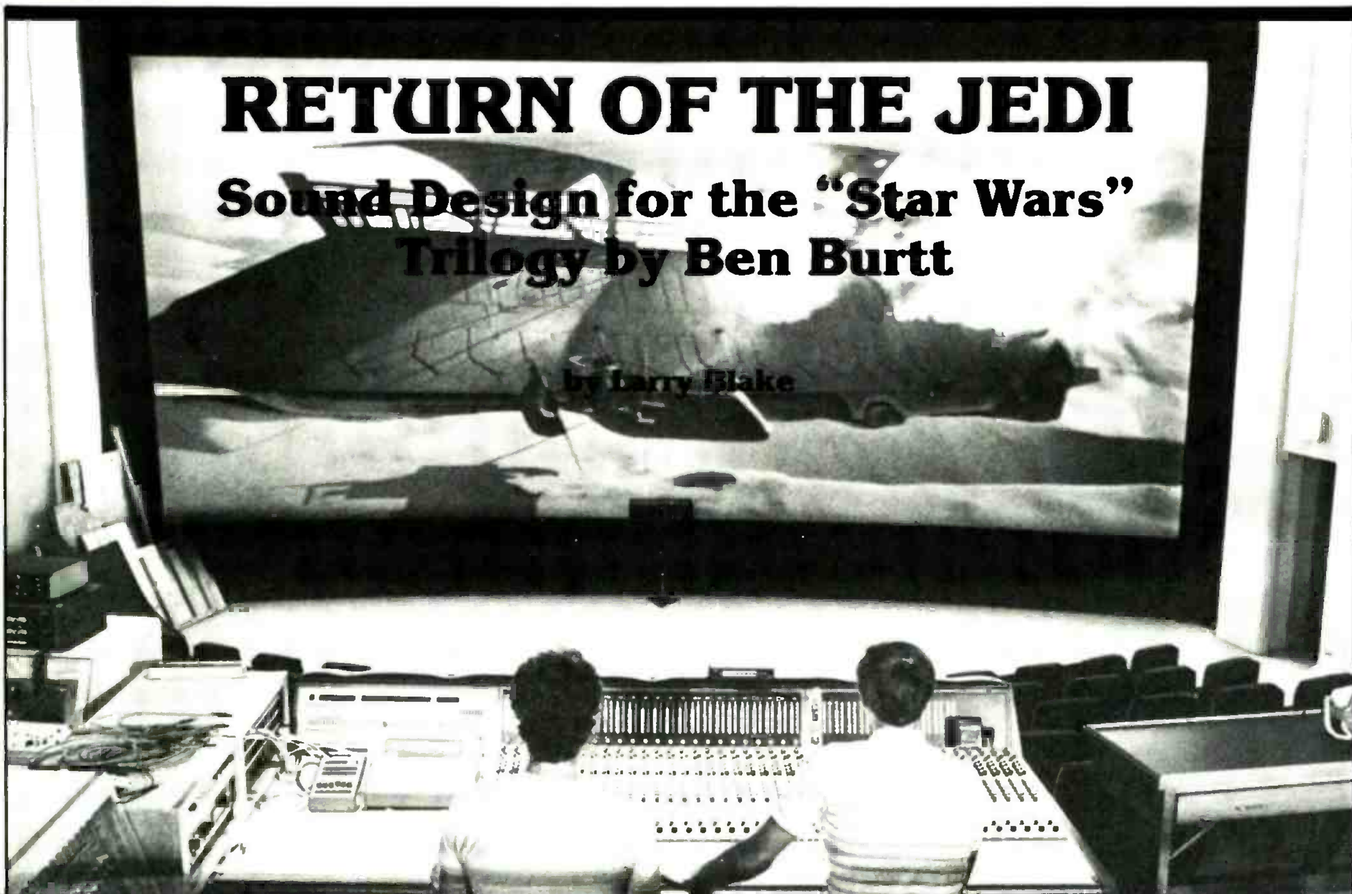
With such a wide range of recording and production environments at a client's disposal, it would appear to be no idle boast when Neuberger says that Chicago Recording "handles more commercials work than any other recording complex in the city. We have four of the six busiest production studios in Chicago, equipped with up to eight-track facilities for voice-over and multi-effects work. Also, CRC records a great deal of original music for radio and television commercials. Unlike other cities around the country, Chicago wasn't as affected by the recent lull in record label work that happened throughout the entire studio industry. Studios in this city have *never* lived or died by record business; the demand for jingles and commercials always has remained healthy."

Next issue: London's cosmopolitan studio community. □□□

RETURN OF THE JEDI

Sound Design for the "Star Wars" Trilogy by Ben Burtt

by Larry Blake



Photography by Barbara Lakin

When *Star Wars* opened in the summer of 1977, the public seemed to embrace not only the movie, but the complete mythological world that it created. The unique soundtrack — ranging from R2-D2's beeps and Chewbacca the Wookiee's grunts to the cracks of the laser swords — made a contribution that undoubtedly was every bit as essential to the film's impact as the revolutionary motion-control effects photography. (Seen on the streets of Los Angeles was a car bumper proudly proclaiming: "I Heard *Star Wars* at the Chinese.")

The sound of the *Star Wars* saga has been guided from the very beginning by Ben Burtt. Writer-director George Lucas and producer Gary Kurtz knew that the soundtrack needed a fresh approach, and in 1975, two years before the first film would open, the duo hired sound effects fanatic Burtt from their film school alma mater, USC.

Burtt spent over a year collecting the sounds needed for *Star Wars*, recording anything that could be turned upside down and backwards to make Lucas' world come alive. Early on, he and Lucas established the style of the soundtrack that would define Burtt's *modus operandi*: "In my first discussion with George [Lucas] about the film, he said — and I concurred with him — that we wanted an 'organic,' as opposed to electronic and artificial, soundtrack. Since we were going to design a visual world that had rust and dents and dirt, we wanted a sound world which had squeaks and motors that may not be smooth-sounding or quiet. Therefore we wanted to draw upon raw material from the real world: real motors, real squeaky door, real animal sounds, real insects; this sort of thing. The basic thing I do in all of these films is to create something that sounds believable to everyone, because it's composed of familiar things that you can't quite recognize immediately.

"You don't know exactly that you're hearing a coffee blender. But the fact that you know what a coffee blender sounds like, and you know it has a motor with this much power, makes the door with a motor seem to be real. You understand from the sound that it must be heavy, and made of iron, etc. The cardboard doors that slide on the [space] ship no longer seem like cardboard; they're solid steel. We emphasize the fact that we record real sounds, and camouflage these familiar items in a way so that the whole *ambience* of a film seems believable."

Burtt's responsibility on *Star Wars* was to create specifically unusual sounds — weapons, vehicles, character and key backgrounds. To do this work he had relatively simple equipment that he "put together with a bunch of RCA cords" in his apartment: TEAC four- and two-track decks, a stereo Nagra, some basic outboard gear, and a 35mm mag transport to make his own transfers. The rest of the sound editing, Foley, and dialog cutting, was handled by sound editor Sam Shaw's crew.

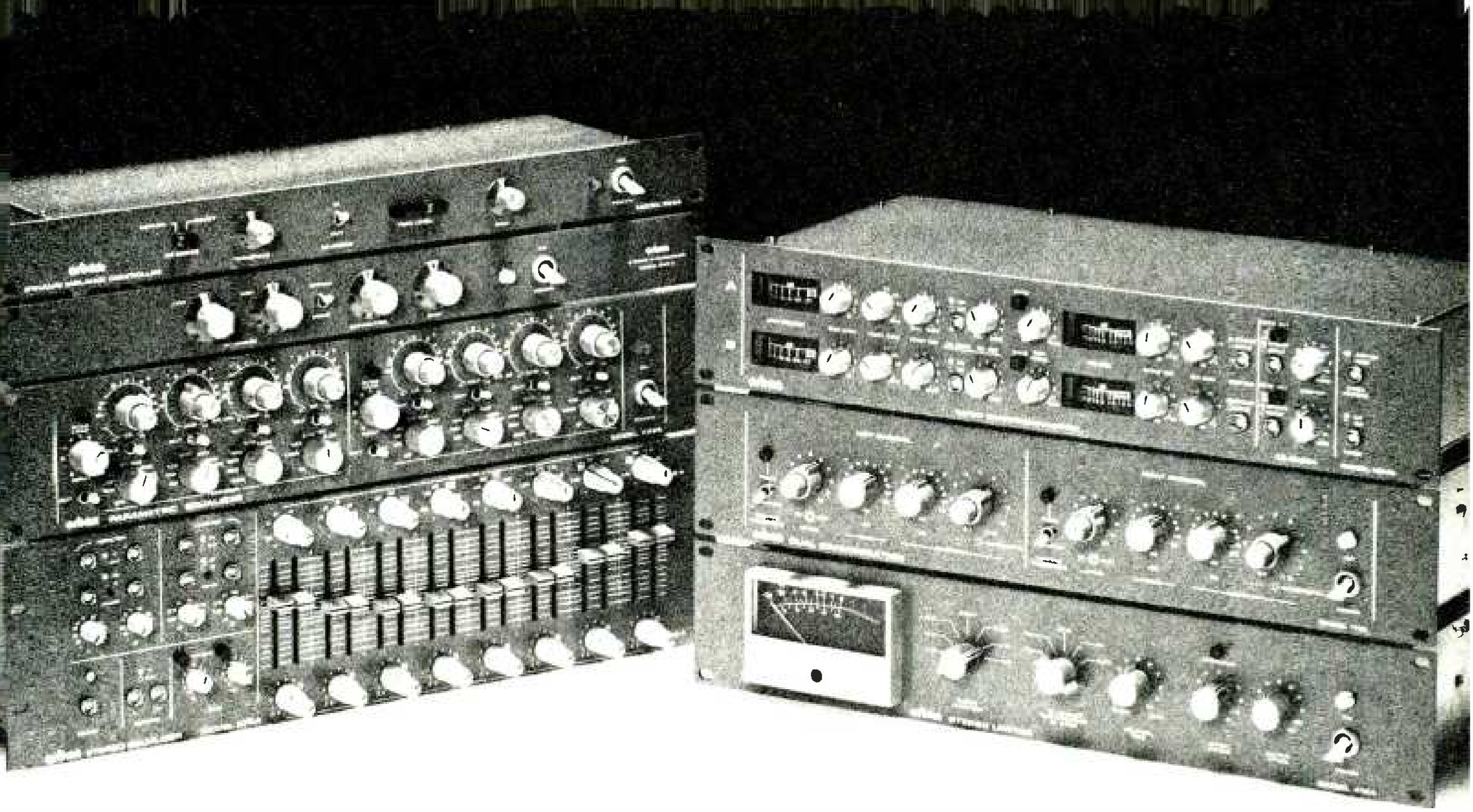
For the second part in the *Star Wars* trilogy, *The Empire Strikes Back*, Burtt supervised both the sound design and all sound editing. The film was mixed by Bill Varney, Steve Maslow, and Gregg Landaker at Samuel Goldwyn Studios, Hollywood, as were two other Burtt sound projects, *More American Graffiti*, and *Raiders of the Lost Ark*. Burtt was also responsible for the voice design of *E.T.*

Chapter VI of the *Star Wars* saga, *Return of the Jedi*, would mark Burtt's first time as re-recording mixer. However, his work began before the production shooting in England, over a year and a half before the film was to open.

Recording Production Sound in the Studio

Ben Burtt went over to England in advance of the shooting of *Return of the Jedi*, to interview prospective production mixers. One of the requests/demands was that the production tracks be recorded with Dolby A-type noise reduction, an idea that many film mixers greeted with "Thanks, but no thanks." "In practice," Burtt says, "most production mixers didn't want to use Dolby A. It added another piece of equipment to the chain."

Finally chosen was Tony Dawe, who came with a good recommendation from Burtt's sound designer friend Dale Strumpell, who had been very pleased with Dawe's production tracks on *Dragonslayer*. (Four weeks of shooting in the U.S. were recorded by Bay Area



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536A Dynamic Sibilance Controller
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mixer Randy Thom.)

All of the stage shooting in England was Dolby A-type encoded, using custom boxes made by Lucasfilm chief engineer Tom Holman. (It should be noted that Dolby Laboratories does not require, or even recommend, that production tracks be encoded, and, indeed, *Jedi* is probably the first major feature to use Dolby noise reduction in production.)

Since this custom box contained Dolby Cat. No. 55 noise reduction cards, access to AC power was needed which, because the units only would be used for stage shooting, should not seem to be a problem. Yet, as Burtt explains, "Believe it or not, there's difficulty in England in getting mains power on the stage for sound equipment. I witnessed it many times. They'll use up all the outlets for lights. There were several times when we couldn't set up because we couldn't plug in the sound equipment on the stage."

The idea behind the use of Dolby noise reduction on production had little to do with reducing tape hiss, since the background level of even the quietest set is still above the noise level of a Nagra. Instead, Dolby encoding of production tapes was looked upon as a way of combating print-through, which sometimes is a major problem with dialog recording. In retrospect though, Tom Holman is not so sure that it was worth the trouble; not because print-through wasn't reduced, but because so little production dialog made it into the finished film.

Among the suggestions made by Burtt and Holman in a memo to the production crew were that, to avoid making noisy footsteps, the actors wear padding on their feet when they couldn't be seen on-camera; that costumes have pouches for radio mikes; and that Sonex acoustic foam and other sound-absorbing material be used to deaden an interior spaceship set to remove unwanted echo; and so on. The latter materials were purchased, but never used.

"There is a tremendous cost, per minute, on the set," offers Burtt. "And, since people know that you can loop a film later [to replace dialog], it is judged more economical not to waste valuable time on the set trying to get usable sound.

"The most 'successful' soundman in the business seems to be the guy who can stay out of the way of the camera crew, and not complain or slow things down. He's kind of off in the corner somewhere with his radio receivers, not causing anyone trouble.

"Sound, you have to realize, has *no* status in production — at least not in the production of our films."

On one filmed sequence, when Burtt got an opportunity to be a boom operator for a day, he had a chance to experience first-hand the problems faced by production sound people. Initially, the scene was shot with two cameras, one of which was set up on such a wide angle that his boom could get no closer than six or seven feet. When the cameras moved in for the close-ups, Burtt thought that here was his chance to get the tracks. On four successive takes he encountered: take #1, a smoke machine making mist *and* noise; take #2, the script girl's watch ticking, plus the director of photography's watch beeping the hour; take #3, the prop man spraying trees right before cameras rolled, resulting in the "plop, plop" of water throughout the take; and, on the fourth and last take, a noisy camera magazine.

While this scene initially was looped, George Lucas liked the original performance of one of the characters, so noise had to be added to the looped character's background to match that of the production track. In the end, music that was intended to begin later in the scene was brought in earlier, because there was nothing else that could be done about the noise.

Part of the production plan was to use Schoeps mikes for dialog, an increasingly popular choice on film sets. However, noise problems on sets were bad enough that the extra reach of a Sennheiser MKH815 shotgun mike often was needed. "Ultimately the goal became just to get an intelligible track. We never had pristine conditions," says Burtt.

Even though all of the non-English creature voices are created in post-production, Burtt found it helpful to give the human beings in the bodies of these creatures some idea of what they would sound like. For example, he played tapes of early sketches of Jabba the Hutt for its three operators, to give them a sense of how the creature would sound in the final movie.

For Burtt the largest amount of new work on *Return of the Jedi* was to create a language for the Ewoks, the inhabitants of the forest moon of Endor that aid the rebels in their fight against the Empire. Since the little creatures had mouths and were often seen in close-up, Burtt had to give the actors a sense of how they should look while speaking.

"I got them together and told them what I'd be doing in post production," he recalls. "I played them some samples of the Ewok voices I had at that time, so they got a sense of the pace of it. I gave them certain instructions in general, and they kind of followed it: not speaking in long paragraphs; getting it over in three or four moves; and doing a lot of body language where you can hide the lack of sync."

During shooting of *Star Wars* and *The Empire Strikes Back*, Chewbacca spoke English, but now he imitates the sound as he thinks it will be in the final movie. The problem is that sometimes his production groans will cover an actor's lines, resulting in the scene having to be looped.

Return of the Jedi director Richard Marquand, a former actor, read Darth Vader's lines from off-camera. On the first two films the lines were read by David Prowse, who is the "body" of Vader.

"C-3PO is generally radio-miked during original shooting," says Burtt, "because he talks a lot, and his movements and body language sync up with a great deal of what he says. It's important to have a reasonably good wild track. Later on, it's re-recorded wild to his original guide track.

"Of course, with a film like this [where we have] several main characters with no mouth, often the dialog gets rewritten to fill in the exposition that's missing. Once the film gets [put] together you always go to 3PO and Vader to clear up the story line. They can say *anything*: 'Yes, Luke, I remember earlier in the film . . .' That literally happens, and is one of the saviors of this type of film."

Sound designer Ben Burtt (left) and Lucasfilm chief engineer Tom Holman.

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Adds re-recording engineer Gary Summers: "I wouldn't be surprised if George [Lucas] had a scene in his back pocket where 3PO and Vader meet and *talk*." (They met only once in the series, briefly, in the carbon-freezing chamber in *Empire*.)

Sound Effects Working Methods and Techniques

Having started recording effects for *Jedi* in 1981, and with principal photography finished, the next step for Ben Burtt was to create sound effects against an early cut of the film. Such an early start — over six months before beginning the final mix — not only gave Burtt the time to do as much as possible by himself, but it also provided the other filmmakers — visual effects supervisors, picture editors and, of course, director and producer — with early versions of the soundtrack to fit to their work.

This work is essential, as Burtt explains: "With this kind of picture you have to wait until you complete a whole mix of a reel before you can even say, 'Does it work as a dramatic piece of film?' You have to get it almost done to be able to say if it works. If it's *Ordinary People*, you can kinda tell in the rough cut if the picture is going to work, because the basic content of the picture is dialog. You may add music, and smooth things out to enhance what is there, but you can basically judge, 'Yeah, dramatically it will work.'

"But if you have a 10-minute sequence with all aliens and opticals — which of course have no sound when shot — you can't even see if it is going to work visually until you've got it in a form where it sounds like a movie."

To judge camera placement and movement before committing themselves to expensive film shooting and optical printing, the staff at Industrial Light and Magic, Lucasfilm's visual effects division, now is making use of the Ultimatte video system. This allows the effects crew to matte together spaceships, planets, etc., using the conventional chroma-key principle. Having a close approximation of what the finished optical will look like gives the picture editor a good idea of what will be needed to cut a scene together.

The premier set piece for both visual and sound effects in *Jedi* is the "speeder bike" chase through the forest of Endor. Burtt's first contact with this scene was through these "videomatics." "Initially, I cut a version which had no finished opticals, and was all videomatics, with ships on sticks flying up against a chroma-key screen," he recalls. "It was a mocked-up version that gave the editors something to cut to. So they can say, 'Okay, we want a shot of it going left to right, lasting so many frames.' Then ILM would actually animate and photograph the real thing.

"George and [picture editor] Duwayne Dunham sat down and cut this surrogate version months before ILM actually shot the scene. I knew that the timings would change for whatever sound I did to that version. But I did cut some sounds, and it gave me a foothold as to what the bikes might sound like. I started out having them sound more like rockets and jet planes but, as it developed, it became more and more like motorcycles with gear shifts and combustion-engine sounds."

Sounds for the 2½-minute bike chase were created and edited by Burtt almost exclusively using a 24-track recorder and Audio Kinetics Q.Lock 3.10 synchronizer in his mix studio. Having assembled a set of work tapes containing sounds that would serve as the starting point of what would eventually become speeder bikes, etc., Burtt would spin effects in sync onto the 24-track using SMPTE timecode on the ¼-inch tape. [For further details, see the accompanying sidebar.]

Each aspect of the sound for this scene — bike pass-bys, bike crashes,

SPROCKET SYSTEMS DUBBING AND POST-PRODUCTION STAGES
Neve Film Re-recording Console with NECAM II and Otari MTR-90 Multitrack Head Up Equipment Complement

The dubbing stage at Sprocket Systems, designed by Jeff Cooper, was built in late 1981/early 1982. The first film to be mixed there was *listen . . .* , produced by Dolby Laboratories and designed to be shown as a short in commercial theaters. Before *Return of the Jedi*, only one other feature film used the new facility: *Twice Upon a Time*, the animated film directed by John Korty and Charles Swenson, and executive produced by George Lucas.

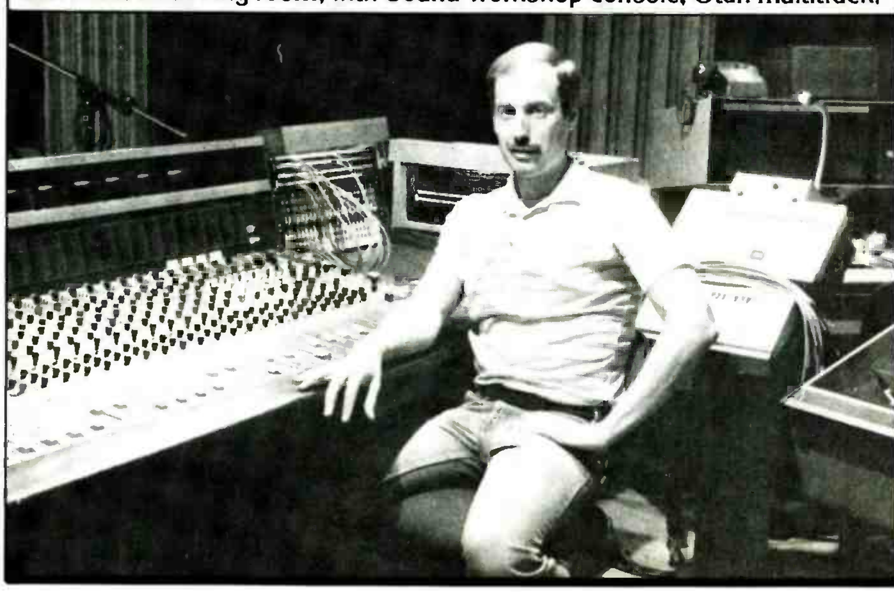
Needless to say, a THX Monitor Loudspeaker System is installed in the room. Inputs to each of the six Boston Acoustics A-200 surround speakers is separately accessible, facilitating any conceivable surround sound experimentation.

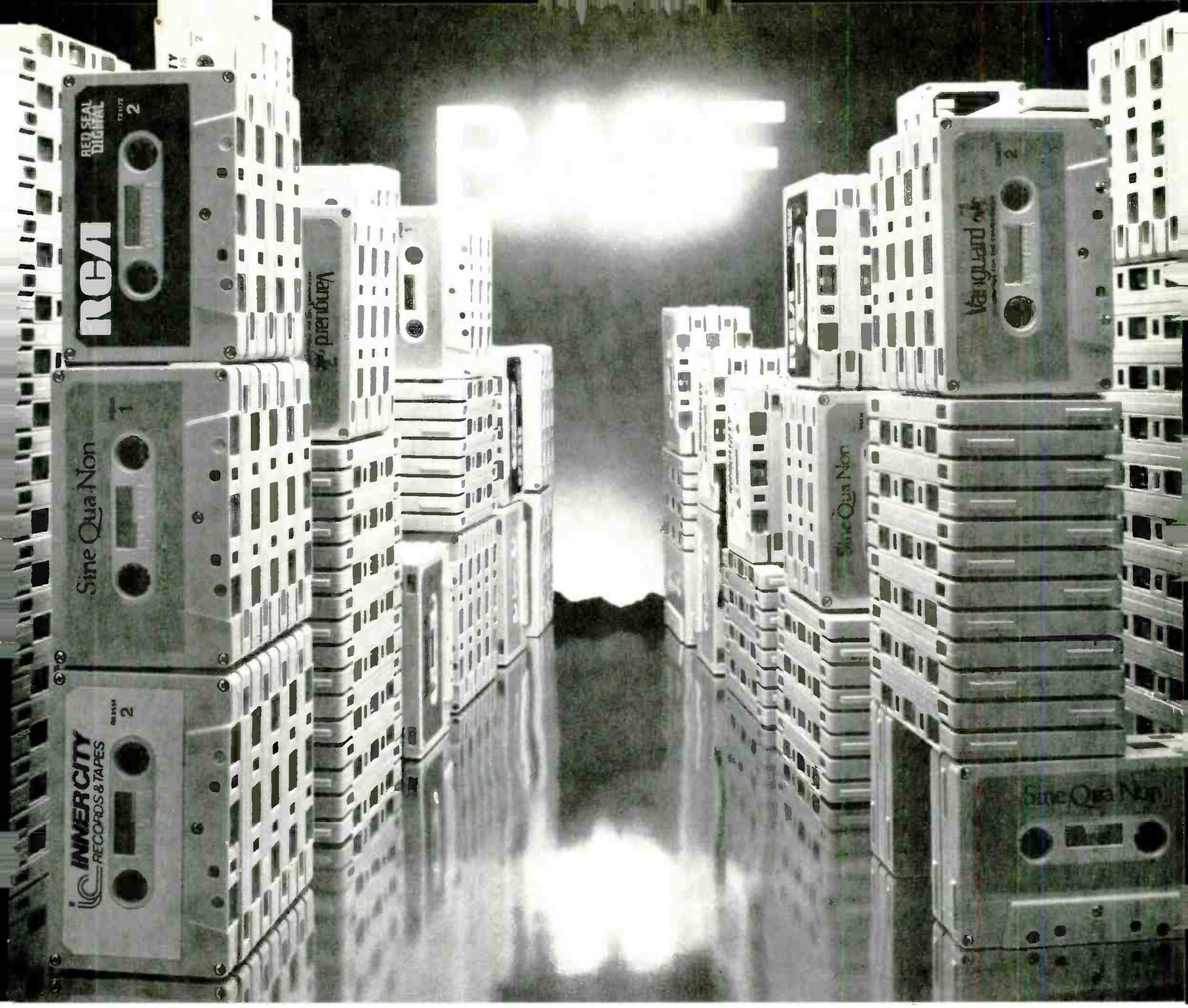
A Neve 8108 32/24 console with NECAM II floppy-disk automation was selected by Holman because of its "quality and flexibility. We are doing many odd configurations of the board that would be impossible with more traditional film re-recording console." This is the first Neve board used in film dubbing in the United States, and also the second use of NECAM II in a film dubbing situation, the first one being at Bavaria Atelier studios in Germany, where *Das Boot* was mixed.

The machine room contains 12 Magna Tech Electronics dubbers for playback of 16mm and 35mm mag elements, with a 13th unit running SMPTE timecode for NECAM synchronization. Two six-track 35mm recorders, an Otari MTR-90 24-track, and 56 channels of Dolby noise reduction round out the primary equipment list.

Holman has worked on, but not had a chance to implement because they have been busy "making movies," a custom pre-amplifier for the playback dubbers that would have head-bump compensation and taloring of the high-end response, to get it "very, very flat." In addition, a separate pre-amp would be assigned to every track format, one each for six-, four-, three-, and single-track heads, making a grand total of 14 pre-amplifier channels. "Machine-room time in a changeover is such an important issue," Holman says. "It would permit us to align the machine electronically, and just mount heads when changing formats, allowing you to roll almost instantly."

Ben Burtt's dubbing room, with Sound Workshop console, Otari multitrack.





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motor noises, etc. — was mixed to a 35mm three-track, left-center-right pre-dub that went to the final mix. Thus, having the sound for the scene on 35mm mag made the editing problems of the 24-track format a moot point, and enabled Burtt to use the multitrack medium where it would help him best. "The main advantage of the 24-track" he says, "is when you are dealing with a scene in the movie for which you are going to invent the sound, and for which there's no precedent. You can add a sound and evaluate it. And add the next layer, and the next, and the next."

Another example of the use of the 24-track occurred at the beginning of the film, when C-3PO and R2-D2 are at the door to Jabba's stronghold, and it opens. "I developed a sound for the whole door as a stereo pre-dub, effectively exactly what we were going to need in the final [mix]. I knew that shot was going to be in the movie. I didn't know the exact footage but, a year later, when the reel was locked, I took that pre-dub and cut it in. Had I had to wait a year with all of the individual tracks, I would never have had time. I might have said, 'Put in a door from the last film.'

"This way, when the giant door opened, I first added a rumble, then a crunching sound, then some squeaking. So, instead of cutting 20 tracks, and then waiting for a month until I got to the stage to see whether I did it right, I can evaluate right then and there, and say, 'Yeah, that's as far as I need to go.'

"It's a correct monitoring environment, decoding through the [Dolby Stereo] matrix, and maybe you even have the dialog and music right there on the 24-track as reference, so you can tell how it's going to sound in context.

"When I get to the stage [at the final dubbing session] maybe 80 to 90% of the decisions have been made, and now it's just a matter of finessing it. For a film like this, when you are inventing so many things, it's an invaluable tool."

Despite its great help with creating complex sound effects, Burtt emphasizes that the 24-track format would not be a practical system for cutting in, for example, door slams or Foley. "If you're working with one sound at a time, and you knew that the sound was going to work [because] it was just a matter of synchronizing it, then I think film is faster, and speed becomes important."

As noted earlier, since all the sound effects in *Jedi* go to the dubbing stage on 35mm mag film, the editing problems of the 24-track format never have to be considered. Nevertheless, the act of recutting the 35mm elements was still a daily problem for Burtt. "When they change the picture cut I'll have to modify, recut or resync," he says. "There's no consideration for sound in the process; the *visuals* come first. We spend a vast percentage of our time resynching and remixing. I'd say certainly 25 to 30% of our time, overall, is spent redoing things, because the picture sync has changed; it might be a cosmetic thing, such as smoothing a bump in the track because something has been cut out."

In keeping with a philosophy set forth at the start in 1975, almost all of the sounds that comprise components of the sound effects used in the *Star Wars* films begin life as live, acoustic events, and not electronic synthesis. However, there are exceptions, as Burtt points out: "Occasionally, for an 'undercurrent' tone in a room I'll combine a tone from a synthesizer with wind [sounds]. I don't think of it first. I'll usually play around with various sounds: air-conditioner backgrounds, or elevator motors slowed down. If it doesn't seem right I might add a tone to fill up an area.

"People are surprised when they come in there, and don't see a lot of electronic equipment and computers. We basically rely on just a Nagra and some microphones to capture the initial raw material."

. . . continued overleaf —

SPROCKET SYSTEMS DUBBING STAGES — continued . . .

Holman felt fortunate in securing the services of Brian Kelley, who previously had been the West Coast representative for Neve, and had installed the Lucasfilm board. Among the modifications that Kelley helped Holman make to the board was enabling the small monitor faders to be used either as additional inputs during a mix (but not under automation control), or to play pre-mixes through the monitors only, facilitating what Holman calls "mixing in context."

Usually, if all 32 main inputs are being used, so are the 12 available dubbers. Which means that if the 24 monitor faders are brought into service, additional elements can be transferred to the facility's Otari MTR-90 24-track.

Holman also modified the 8108's panpots: "It's not a standard sine/cosine panpot. Ours is modified to work well with the Dolby matrix, and has certain bumps in it. Actually, it is only 2/3 modified for the matrix; otherwise we would compromise the pans in the discrete mode too much."

Ben Burtt's Personal-Use Dubbing Room

Perhaps the most unique part of the Sprockets System facility is the small dubbing room that serves as a "high-tech Moviola" for Ben Burtt. The heart of the system is an Audio Kinetics 3.10 Q.Lock synchronizer featuring custom software modifications written by Steve Waldman of Screen Sound in L.A. [now president of Audio Kinetics U.S.] to Burtt's specifications and requirements. As he says, Burtt was looking for an answer to a long-time desire: "A KEM with 24 sound heads that I could use to make evaluations of what I was working on." [According to Waldman, the original KEMulator software forms the basis for Audio Kinetics' current Q.Soft SFX software — *Ed.*]

Rather than locking a videocassette player to a 24-track recorder, as is standard video sweetening practice, Burtt's system involves interfacing a film chain, including 35mm projector and mag recorder, along with, in various combinations, ¼-inch SMPTE timecoded tape, 35mm mag elements on three film dubbers, and a 24-track tape recorder.

In fact, the only item Burtt does *not* lock up to is a videocassette recorder, since he uses a black and white 35mm "dirty dupe" of the editor's color workprint. Using standard 35mm projection allows him to see and hear the correct relationship between stereo sound and wide-screen picture. Burtt further simulates theatrical presentation by pressing a button on his console that places his four-track mix through the Dolby Stereo matrix.

The ¼-inch timecode system used at Lucasfilm was devised by Tom Holman and Howie Hammermann in 1981 and, as it turns out, has almost identical track width and SMPTE timecode level to the system used by Studer in its new A810 recorder. "We both picked as high a level as we could get away with," notes Holman. [Studer] did it because they wanted to be able to run at 1/20 speed and be able to detect it. Our biggest problem is that Ben has a lot of recordings of thunder crashes and things like that which have a *lot* of bass crosstalk, head-to-head, [which] gets into the timecode channel and screws up the code."

One difference between the two systems is that Lucasfilm does not use a delay line to sync the output from the timecode head with the record and playback heads. The reason is that the Lucasfilm library bears no relationship to picture; timecode only is used to run the ¼-inch tape along with the film chain for synchronization. Because of this, the timecode is applied to the sound effects tapes *after* they are recorded, thus doing away with the need for a timecode generator in the field. Practically all sound effects recordings at Lucasfilm are made with Dolby A-type noise reduction, as are all transfers to ¼-inch and to 35mm mag. □□□

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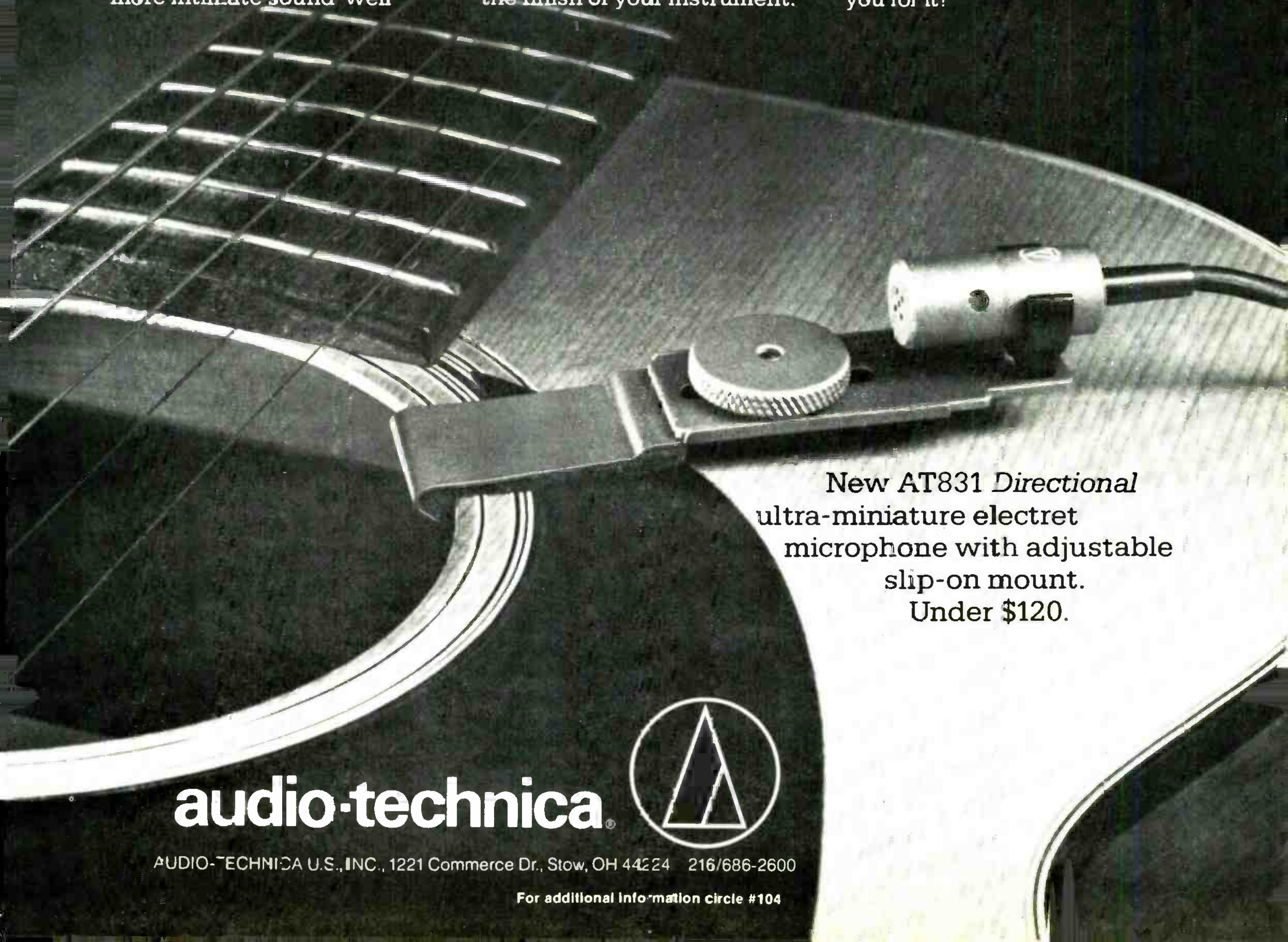

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Creature Creation

Ben Burtt undoubtedly is very happy that only a minute percentage of the six million languages that C-3PO speaks have been heard so far in the *Star Wars* saga. The most work that he had, language-wise, on *Jedi* was the creation of the Ewokese language. Here he describes some basic components of their language, plus general guidelines he follows when creating a new tongue: "I started by recording the Tibetan, Mongolian, and Nepali languages. I broke the sounds down phonetically, and re-edited them together to make composite words and sentences. I would always use a fair amount of the actual languages, combined with purely made-up words. With a new language, the most important goal is to create emotional clarity.

"Most of the attempts I do at making a voice I try to generate in front of a microphone. There might be some kind of futzing of it later, but usually the most successful one is performed live with a minimum of processing.

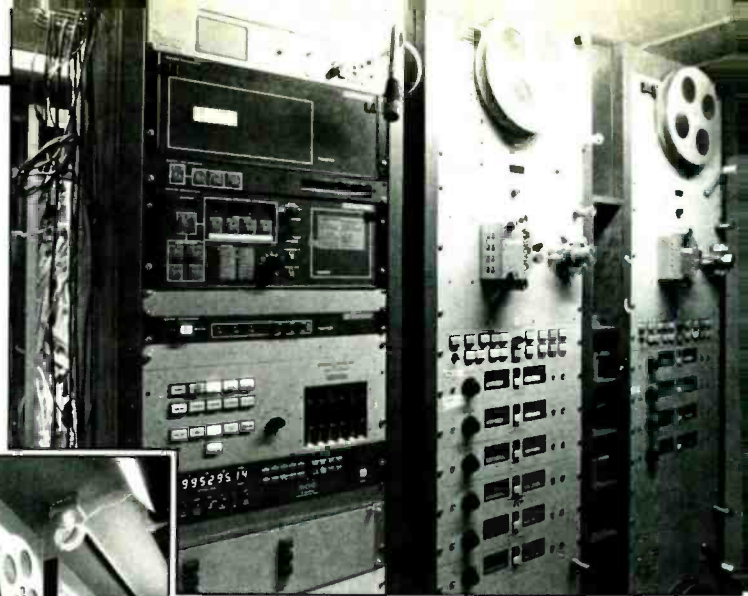
"People spend all of their lives learning to identify voices. You become an expert at that, whether you know it or not. It's very difficult, and somewhat impossible, to electronically process the human voice and eliminate the human characteristic, and retain the necessary emotion. To fool the audience into believing this is a real character, you kind of have to have one character as the basis of the sound, although you may sprinkle other things in there. There's no formula. It varies from character to character.

"The general process is one of a very few number of tracks, with maximum effort at selecting the individual sound and recording them in the first place. It's not a difficult mixing process; it's more of an editorial process."

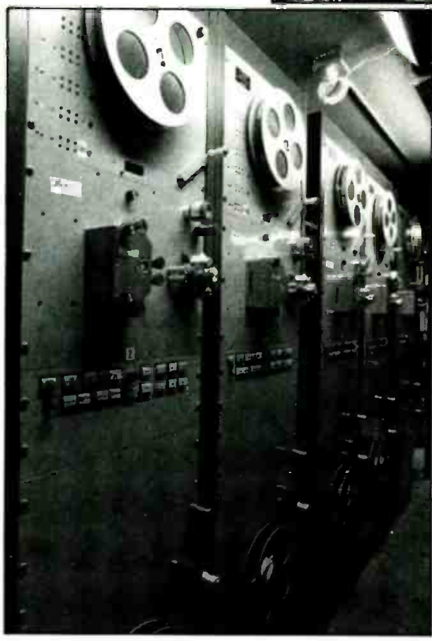
The other big challenge for Burtt on *Jedi* was the "vile gangster," Jabba the Hutt, who had Han Solo frozen in carbonite for non-payment of debt. (This, presumably, is what was done "a long time ago, far, far away," instead of giving someone "cement shoes." Jabba had actually been filmed for *Star Wars*, but Lucas was not pleased with his appearance, and cut his scene.)

In *Star Wars*, audiences did get a taste of his language — Huttese — as spoken by the hit man, Greedo. Burtt worked with Larry Ward, then a student at UC Berkeley, near San Francisco, in devising Huttese; Ward, who speaks many languages, was the voice of greedo and, later, Jabba.

Burtt and Ward phonetically wrote out lines for the latter to speak, matching the production track of the puppeteer who had said Jabba's lines in English during production. Dialog coordinator Laurel Ladevich cut these lines



Lucasfilm machine room houses two Magna-Tech 35mm mag 4/6-track recorders, and a Dolby CP-200 cinema processor. Below: Five of the 13 Magna-Tech 35mm playback dubbers.



into time-honored loops, with beeps leading up to a line, for which Ward would then try to improvise a syllable-for-syllable match. Both versions were recorded on ¼-inch tape for later selection and transfer to 35mm. "Much of what Jabba said may have been contracted from pieces of things Larry Ward said over a period of six or seven months," recalls Ladevich. "Providing that the recording quality matched, I would put them into one sentence."

This replacement work originally was done in the summer of 1982, and used for a temp dub recorded in July. Later recutting of scenes necessitated bringing back Ward for more work on Jabba. "Not only was this expensive and time-consuming," Ladevich notes, "but everyone had become attached to the way Jabba was in the temp mix."

Other notable vocal contributions were made by Eric Bauersfield, who dubbed Admiral Ackbar and Bibb Fortuna (the creature with the snake growing out of his head), and Kip Sang Rotich, who played Nien Nubb, Lando Calrissian's co-pilot in the space battle. Most of what comes out of Nien Nubb's mouth is Kikuyu, Rotich's native tongue back in Kenya. Burtt had Rotich pretend he was a commander leading his troops — which presumably were not battling in space — and put parts of the story in Nien Nubb's mouth.

Just as Burtt connects sound effects to reality using actual recordings, often the reality "hook" of a language comes not from a part of an existing language, but from a sprinkling of pidgin English here and there, as when Bibb Fortuna said "Bargon no wachonga" — which, of course, means "There will be no bargain." (Students of such wordplay should see *The Dove*, the hysterical short parody of Ingmar Bergman's films. Most of the words that people speak are English, with "-ska" added. For example, "Peachy black nightska" was subtitled as "Darkness.")

For security reasons, none of the people who voiced *Jedi* characters saw any footage from the film. Laurel Ladevich believes that this works well, because "when you're dealing with people who don't loop every day, and you showed them a picture, and had a track, and they had something to say, it would be too much. People like Harrison Ford, Carrie Fisher, and Mark Hamill are familiar with looping, and when they go on the stage it doesn't intimidate them to see a great big picture of themselves."

As Burtt's responsibilities have increased over the three *Star Wars* films, so has the style of the sound been established. This allows him to pick and choose the areas to cut personally, and to leave the remainder to his staff of editors. "I basically roam around doing the things I know I can do best," he says, "and that I couldn't explain easily to anyone else. My style and technique is already reflected in the library. I'll check and modify and sweeten, but the bulk of the work goes to someone else. One exception is R2-D2. I could farm him out, but I have always done R2, and reserve that one thing for myself."

Sounds for R2-D2 also are an exception to Burtt's general rule against the use of synthesizers, since about 50% of the droid's voice is generated electronically; the rest is a combination of water pipes, whistles, and vocalizations by Burtt. Tried, but rejected because of its *over* familiarity to movie audiences, was a Touch-Tone phone.

Like most of the creatures, R2-D2's

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Introducing the RE30 and RE34.

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An LED, mounted so as to be easily visible to the talent only, serves several important functions...it shows the presence of phantom power, monitors battery condition, and offers the world's first hand-held "tally light" to signal on-air personalities from off-camera.

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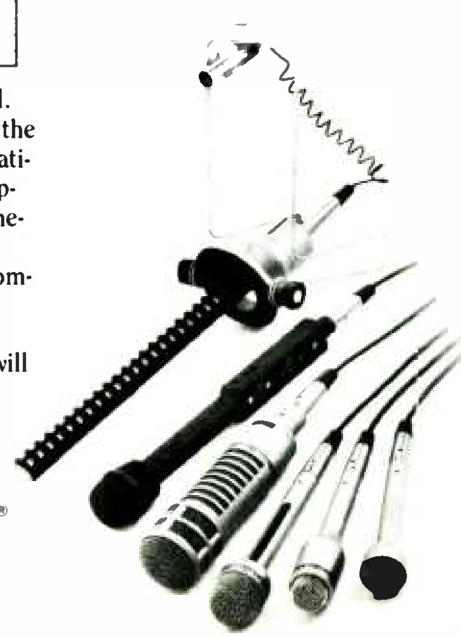

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Executive producer George Lucas, "Return of the Jedi" cast, and director Richard Marquand.

sound is cut on 35mm mag. In the spring of 1982, Burtt spent a month cutting the effects: "First I will make up a sentence and word lists on four- or eight-track tape. I will then do variations of that, transfer it all to film, and cut pieces out to fit the timing of the scene. R2 is usually no more than one or two tracks.

"I cut R2 on film, because it is the kind of process where you are going back and forth between actor's lines, and timing is crucial. It's linear, and doesn't relate to anything but the dialog track, which I run along on the KEM. The 24-track system is not as efficient for a sound which is one linear sound all by itself."

One component of R2-D2 that Burtt doesn't cut are his motors, which are cut by Terry Eckton, who also cut Chewbacca the Wookiee for *Jedi*. R2 and C-3-PO's motors take two people between six and seven weeks to cut, covering every move. "They get buried most of the time," notes Burtt. "But when they do surface, it helps keep a consistent texture that tells you this is really a robot.

"In case of things like the Wookiee —something where you're constructing the voice out of pieces of animal sounds — that's a whole different matter. You have bits and fragments of animal sounds which you've collected and put into lists: here's an affectionate sound, here's an angry sound and, just like with R2, they're clipped together and blended. With a Wookiee, you might end up with five or six tracks, sometimes, to get the flow of the sentence."

Music Recording

John Williams' music for all three *Star Wars* films has been recorded by Eric Tomlinson in England. Music for both *Star Wars* and *The Empire Strikes Back* were recorded at the Anvil Studios in Denham, near London. Since that time Anvil has been based at Abbey Road Studios, in north London.

Tomlinson's standard orchestral setup employs three Schoeps omnidirectional as primary feed, plus additional spot microphones on each session. Although a three-track monitor mix is recorded on 35mm mag during each session, what is heard in all three films has been mixed from the 24-track master.

Observant viewers might notice that the speeder-bike chase through Endor has no music, which is notable because the *Star Wars* films have acquired a reputation for wall-to-wall music, especially in battle scenes. During the "spotting" of the film, when George Lucas, John Williams, and Ben Burtt ran the rough cut of the film on a KEM editing machine to discuss where the music should go, and where it should not, Burtt suggested that there be no music during that scene, because he felt "the intensity of the scene would be more pronounced if we surprised the audience with just a point of view reality of visceral bike sounds. I felt it was unnecessary to have music tell the audience that it was exciting. Johnny Williams agreed and George threw up his hands and said, 'Okay, if you guys say so.' "

In those instances when the music is ready, and cut some time in advance of the final dub, Burtt uses his 24-track system to good advantage. "In *Raiders of the Lost Ark* we had the whole first reel of music ahead of time," he explains, "so the whole thing was built up on a 24-track, with the music there as a bed. You would know what to put where, and you'd stop putting in a rumble if you knew that it wouldn't read anymore, or the music was carrying enough of the emotion. The first reel of *Raiders* was mixed in one afternoon at Goldwyn. Three passes: two rehearsals and a take. It was a very complicated reel, and we might have spent a long time on it otherwise."

In case anyone thought that the projectionist had the fader too low when *Return of the Jedi* begins, it should be noted that the Twentieth Century-Fox fanfare at the beginning of the film was recorded at a low level on purpose, to make the opening sting of John Williams' *Star Wars* theme pop out of the speakers.

* * *

In part two of this article, to be published in the December issue, Larry Blake will continue with his detailed description of the sound design for *Return of the Jedi*, including stereo dialog, music and multiple effects for the final mix; surround and boom channel information; problems encountered with the restricted dynamic range of movie theaters; and foreign-language dubs. Also included in the conclusion to this article will be an extensive sidebar detailing the THX loudspeaker system for movie theaters, designed by Lucasfilm's chief engineer, Tom Holman. ■■■

DIGITAL AUDIO PROCESSING DEVELOPMENTS AT LUCASFILM

Having been in use since George Eastman first began manufacturing film for Thomas Edison in the 1890s, the 35mm four-perforations-per-frame format is still the worldwide standard for motion picture production and exhibition. Simply put, every 35mm projector in the world can play back *any* film produced anywhere in the past 70 years, a fact which earns the four-perf format a place next to the Philips Compact Cassette, and the 33-1/3 LP in the "Standardization Hall of Fame."

The reason for this longevity is clear: synchronization is a very simple, mechanical process that has changed very little equipment-wise since the coming of sound.

However, in the real world you don't get something for nothing, and the 35mm film format (like all other gauges) trades simplicity for speed and flexibility. An obvious comparison can be made between the on-line editing of one-inch videotape masters, and its equivalent in the film world: negative matching, cutting and color balancing; and printing, processing and viewing the first answer print. But what may take only a few hours in an on-line editing suite takes days in a film laboratory. This is not to mention the greatly increased flexibility of the video process, which translates to the ability to change one's mind at the last moment.

In a 1981 interview in the magazine *Film Comment*, George Lucas spoke for filmmakers worldwide when he summed up his reasons for funding the use of electronic technology in motion pictures: "Anybody who's worked with film, especially anybody who has edited film, realizes what a stupid, 19th Century idea film is. Anybody who has torn sprocket holes, or tried to show a first rough cut, knows the only thing they worry about — they don't even see the movie — is if the film's going to break. I mean, it's just not what you'd call a sophisticated setup."

To paraphrase Mark Twain, everybody knows this, but only George Lucas seems to be doing something about it. For the past three years, the Sprocket Systems division of Lucasfilm Ltd. has been researching and developing the use of computers in all aspects of filmmaking: picture and sound editing, sound mixing, and film printing.

The computer group is headed by Ed Catmull, with separate projects researching video editing (led by Ralph Guggenheim), high-resolution laser scanning (a.k.a. digital film printer) (David DiFrancisco), and computer games (Peter Langston). The largest department is the graphics division, led by Alvy Ray Smith, and some of its work already has been seen in *Star Trek II: The Wrath of Kahn*, and *Return of the Jedi*.

The digital sound project has been led since its inception in 1980 by Andy Moorer, who was the co-founder of the Center for Computer Research in Music and Acoustics at Stanford. [See *R-e/p* December 1978 issue — *Ed.*] Moorer and his staff of four are putting the finished touches on their prototype ASP (Audio Signal Processor), which they define as "a large special-purpose computing device for digitized audio." The ASP uses the UNIX operating system, running Lucasfilm FMX (Film MiX) program software, which was written by Curtis Abbott. Working with Moorer, who does hardware design and system support documentation, and Abbott, who is chief programmer, are: Alan Marr, user interface programming; Jim Lawson, UNIX system support; and John Snell, console hardware design.

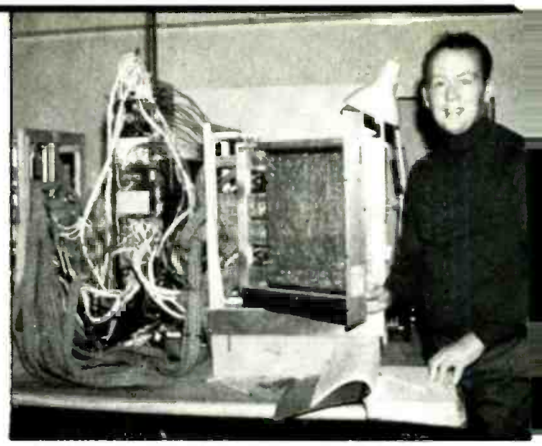
The first software to be written for the ASP concerned the creation of sound effects, to aid Ben Burtt in concocting his aural worlds. However, the experience of mixing *Return of the Jedi* in-house under extreme time pressures has led to the recutting software receiving higher priority than originally intended. "Re-cutting for the digital system is just editing a text file, and Ben feels that the system as it stands now is better in some cases than 35mm, in terms of meeting the demands of changing the sound elements to conform to picture changes," says Moorer.

Hand-in-hand with the problem of recutting is the appearance at the final mix of new (i.e., not pre-mixed) sound effects. Such "sweetener" tracks mean extra inputs are needed to accommodate these late changes. With this in mind, Moorer now expects the number of tracks that can be processed at one time to extend to 64. This figure should be distinguished from the number of channel faders, which Moorer hopes to keep to a maximum of 32.

The cleaning up of production dialog tracks also has been the subject of much post-*Jedi* attention. At this point in their research, Moorer is expecting speech synthesis to be "the real solution" to the dialog clean-up problem. This would entail sampling and analyzing a segment of production dialog, and then synthesizing the voice *sans* noise. To facilitate this procedure, Moorer would like to see some production recordings made with two microphones — one capturing the dialog *plus* the noise, and the other just the noise.

Digital Storage of Film Sound Elements

Until recently, Moorer has been using standard 300-megabyte disks for all storage, playback, and recording. Since their capacity is similar to that of a 1,000-foot reel of 35mm mag film — each disk only stores 10 minutes of four-track sound (or 40 minutes of mono sound) — there would probably be a 1:1 ratio between the reels of 35mm mag film that are used today, and the disk packs needed for such a system. Thus, using such a disk-based



Andy Moorer with prototype ASP.

system, the sound for a film would have to be stored on scores of packs, which would need to be mounted and dismounted during editing and mixing.

However, as any personal computer owner knows, the cost of mass storage continues to fall, as the capacity increases, and this Fall Moorer is looking forward to receiving two demonstrator units of a new disk drive that holds almost three times the data, and which are 50% faster, than the drives currently in use. (The figures work out to be 110 minutes of mono sound, or 18 minutes of six-track.) Also, as might be expected, the new units are much smaller: they take up only 10 inches of 19-inch rack space, while the previous drives were roughly the size of a 24-track recorder.

The only drawback presented by the new technology is that the storage medium is fixed, and cannot be removed. Therefore, dozens of drives would have to be dedicated to all the dialog, music, and sound effects required during the post-production of a feature — a very expensive proposition, even for Lucasfilm.

An obvious solution appeared this year in the form of the 24-track, half-inch digital recording format, which at this present time is represented only by the Sony PCM-3324. The storage media costs are particularly attractive to Moorer: at the 48 kHz sampling frequency, a 14-inch reel of half-inch tape stores 60 minutes of 24-track sound. To give a sense of perspective, as few as six reels could store all the elements required for a complex stereo mix — instead of the 200, 1,000-foot 35mm mag rolls that might be used today. The money saved by not having to align 35mm dubbers with pink noise and Dolby tones would alone be substantial.

Use of a digital multitrack recorder would entail transferring the material needed for that day to random-access disk drives, which then serves as a temporary working store. At night, the day's work would be recorded onto 24-track digital tape(s) in preparation for the final mix.

A third digital storage medium — the recently introduced Compact Disc — is being considered to store Lucasfilm's extensive sound effects library. Instead of reels of ¼-inch tape occupying a whole wall, everything will fit neatly onto one shelf crammed with about 100 Compact Discs. Thus, all sound editors at the facility will have the entire sound library at their fingertips.

Current plans at Lucasfilm call for the ASP to use the current professional standard sampling frequency of 48 kHz. As was reported in *R-e/p* last year [October 1982 issue], describing developments in digital sound for motion pictures, Moorer earlier had thought that 35 kHz might be a sufficiently high sampling frequency, because information above 15 kHz is of negligible importance to theatrical presentation. However, he notes that since "the new disks are so much faster and larger, there is less and less reason to go to something lower [than 48 kHz] to 'conserve' space."

For dialog processing and clean-up that will involve "huge, grotesque amounts of processing, things that would keep a Cray ['supercomputer' costing $7.5 million each] occupied for a year," Moorer anticipates using sampling rates as low as 25 kHz. "The dialog [in these instances] would already be so awful, and there will already have been so much damage to the sound, that anything we would do to it would help."

Initial Installations of ASP System

A modest version of the ASP will be installed next to the Neve 8108 film console on the Sprocket Systems dubbing stage, which location will serve as the testing ground for its use as a mixing station. It should be noted that the ASP will only be used for specific tasks, such as dialog clean-up and effects creation; the 35mm mag format will continue to be used for all editing and mixing. The first film to utilize the digital audio processor, albeit in a limited manner, will be *Indiana Jones and the Temple of Doom*, the sequel to *Raiders of the Lost Ark*.

The complete system will be available for purchase, with delivery of functioning systems expected to begin around Christmas 1984, in-house use having begun six months earlier. Moorer is quick to note that, if one considers "the whole environment" — mixing desks, automation, patch bays, outboard equipment, recording and playback units, etc. — the Lucasfilm ASP will be able to compete with a standard all-analog film dubbing studio on a "dollar-for-dollar basis."

He foresees sale of the system to a new generation of "technocrat" production facilities serving the independent producer, and is especially pleased that, as a result, low-budget movies will be able to afford cleaner sound.

It should be obvious that such a system would reduce greatly the number of people needed to produce a soundtrack by blurring the distinctions between the auditioning, transferring, editing, and mixing of sounds. Thus, with less personnel able to do more work, the "time is money" aphorism will, hopefully, no longer prevent any filmmaker from obtaining a clean, creative stereo soundtrack.

Digital Music Synthesis Applications

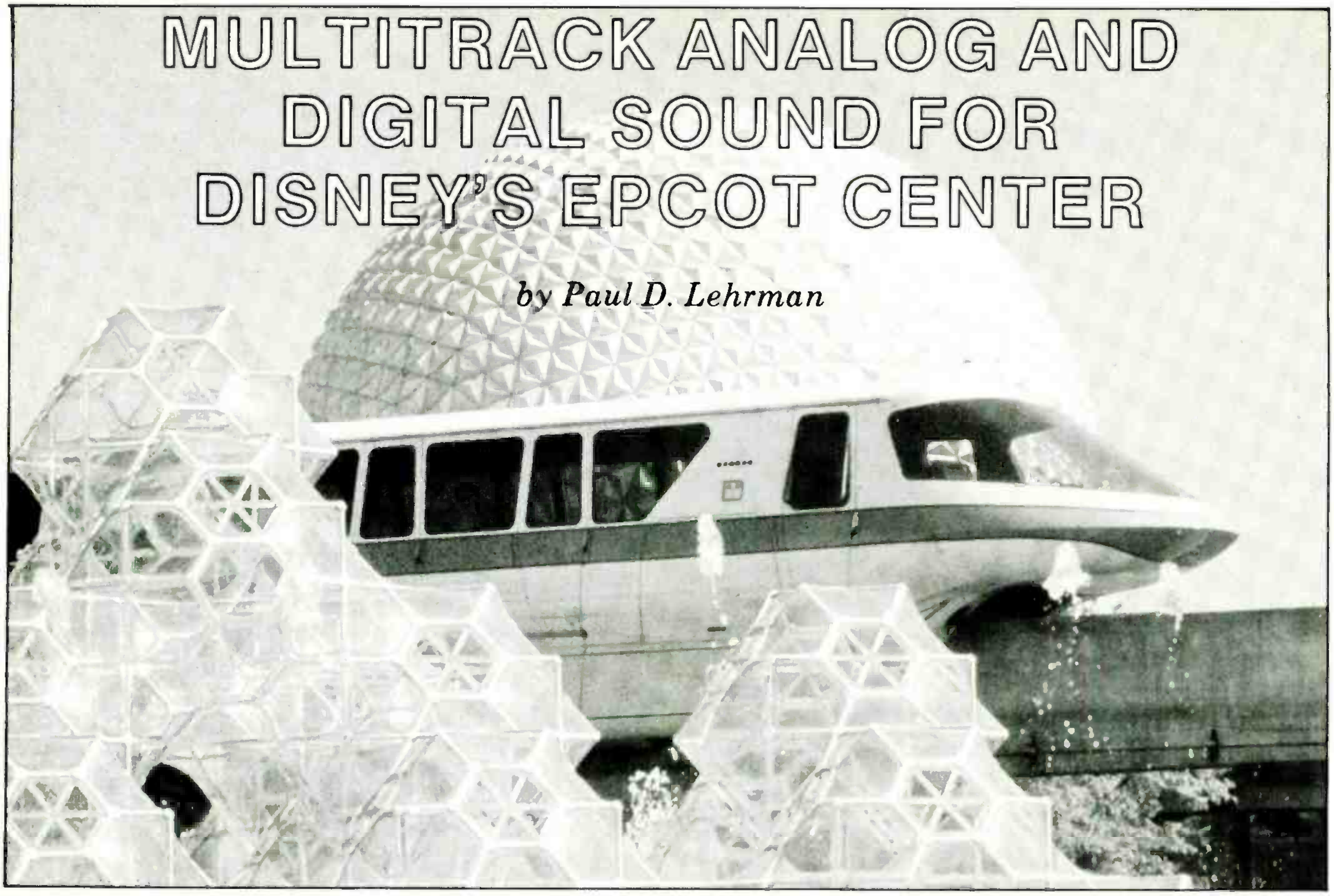
The 35-second trailer which announces the installation of the Lucasfilm THX loudspeaker system at a theater also marks the uncredited public debut, on the soundtrack, of the Lucasfilm ASP. Composed by Andy Moorer, the music for the trailer consists of 30 synthetic "instruments" playing simultaneously in real time, a task far beyond the reach of today's analog synthesizers.

Moorer had a perfect chance to demonstrate the flexible editing capabilities of the ASP when the picture cut was changed at the last moment. It took him all of 10 minutes to re-cut his score, something that might have taken hours had the music been recorded on 35mm magnetic film. Such speed will be a godsend on feature-length films, not only in the editing of finished, synthesized scores like this one, but in the creation of standard symphonic underscore, such as John Williams' music for the *Star Wars* film.

Instead of writing out what amounts to little more than a piano score, and giving it to an orchestrator, all working under the gun to meet a last-minute deadline, computer-aided manuscript editing will allow the composer to *hear* different trial orchestrations via synthesis. This perhaps was possible in the days when each studio had its own orchestra, but those days are long gone. In the end though, Moorer expects live orchestras to continue to be used for the added "oomph" that only human beings can give. □□□

MULTITRACK ANALOG AND DIGITAL SOUND FOR DISNEY'S EPCOT CENTER

by Paul D. Lehrman



The most sophisticated audio/video playback system in the world is not located in New York or Hollywood. It's in Florida, just south of Orlando, at the new EPCOT (Experimental Prototype Community of Tomorrow) Center at Walt Disney World. Visitors to the 260-acre complex are constantly bombarded by sound and images — from soaring pseudo-Korngold movie music, to the bloops and bleeps of a "How-it-Works" computer show; from full-color 3-D movies, to a stair-climbing Audio-Animatronics figure of Ben Franklin.

But the assault on the senses here is very different from the random cacophony of a typical amusement park. Instead, everything the visitor sees and hears is carefully predetermined and controlled by a large network of computers called EPCOT Central, located in a building in the Future World section of the park known as Communicore East.

There are 16 major structures at EPCOT (with several more planned for the near future), divided into two areas: Future World, whose buildings cluster around a small area close to the entrance gate; and World Showcase, which surrounds a 40-acre lagoon. Each of the buildings contains from one to several dozen audio/visual presentations, which range in complexity from simple audio playback systems, to elaborate multimedia theatrical events.

Program Sources

It doesn't take long before the visitor realizes that very little of the entertainment to be had here is live — everything, from the "actors" at American Adventure, to the bird calls at Canada, is canned. But *what* a can it is. At the heart of this audio/video extravaganza is a custom mainframe computer made by Sperry. The computer is surrounded by, and controls, literally hundreds of video and audio playback units at EPCOT Central, while its electronic tentacles reach out to control dozens more devices spread out around the grounds. There is an astounding amount of program material stored on various media at EPCOT Central. (Although there is a Neve- and Ampex-equipped production studio in place at the nearby Magic Kingdom, all of EPCOT's program material was produced elsewhere, as detailed in an accompanying sidebar.)

There are 22 channels of background music, each emanating from its own L.J. Scully 14-inch playback deck, and routed to speakers concealed in the grounds, trees, and buildings via a 70-volt distribution system. Each area of the park has its own background music — 30 minutes of it — which was composed and recorded expressly for this application.

Within each of the buildings, especially at Future World, are brief displays using animated figures, touch-sensitive video screens, and other active and interactive media. Regardless of their location, most of the displays are fed by equipment at EPCOT Central, consisting of five-deck audio cartridge units, Sony interactive videodisk players, and shuttling Sony U-Matic VCRs. There also are still stores, and graphics generators.

Many of these smaller exhibits are much more complex than the "push-a-button-and-see-a-film" displays typical of a Worlds' Fair, or a science museum. In the Future Choice Theater, for example, 170 people sit in chairs whose arms contain five membrane buttons. A live host, standing on a platform underneath a video-projection screen, explains to the audience that they are to use the buttons to respond to what they see and hear during the presentation. Then a number of national and international issues are presented and discussed, using prerecorded interviews and newsclips on videodisk, and the audience is asked its opinion on each issue, in the form of multiple-choice questions. The response from the armchair buttons is instantly tabulated by the computer, and the results displayed on the video screen.

Running the shows is just part of the computer's function. It also is responsible for managing environmental, energy, and communications systems in the park. Among the last are several "WordKey Information Services" kiosks scattered about the park that provide visitors information on exhibits, services, and activities throughout Walt Disney World, using touch-sensitive video screens, and two-way audio/video systems.

All told there are 108 movie projectors

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There are other microphone alternatives when high sound pressure is a factor.

As Sennheiser claims, the MD 421 undoubtedly stands up to extremely high decibel levels and has other features that have contributed to its popularity. But if you're already using the MD 421 to mike loud instruments or voices, we suggest that you investigate the Beyer M 88.

The Beyer Dynamic M 88's frequency response (30 to 20,000 Hz) enhances your ability to capture the true personality (including exaggerated transients) of bass drums, amplified instruments and self-indulgent lead vocalists.

The Beyer M 88 features a matte black, chromium-plated brass case for the ultimate in structural integrity. Beyer microphones are designed for specific recording and sound reinforcement applications.

When you need a rugged and versatile microphone, consider the alternatives.

For over 10 years, engineers have used mics like Shure's SM57 for the widest variety of applications in the studio. And we feel that one of the main reasons more engineers don't use the Beyer M 201 in this context is simply because they don't know about it. Those who have tried it in the full gamut of recording situations have discovered how it can distinguish itself when miking anything from vocals to acoustic guitar to tom toms.

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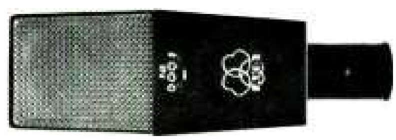
Each Beyer Dynamic microphone has its own custom-designed element to optimize the mic's performance for its intended use.

You may not always need a condenser microphone for "critical" recording applications.



Some engineers prefer condenser microphones like the AKG C 414 to accurately capture the subtle nuances of a violin or acoustic piano. But should you have to deal with the complexity of a condenser system every time this kind of situation comes up?

The Beyer Dynamic M 160 features a double-ribbon element for the unique transparency of sound image that ribbon mics are known for. While its performance is comparable to the finest condenser microphones, the M 160's compact size and ingenious design offers significant practical advantages for critical applications.

Beyer Dynamic microphones offer state-of-the-design technology and precision German craftsmanship for the full spectrum of recording and sound reinforcement applications.

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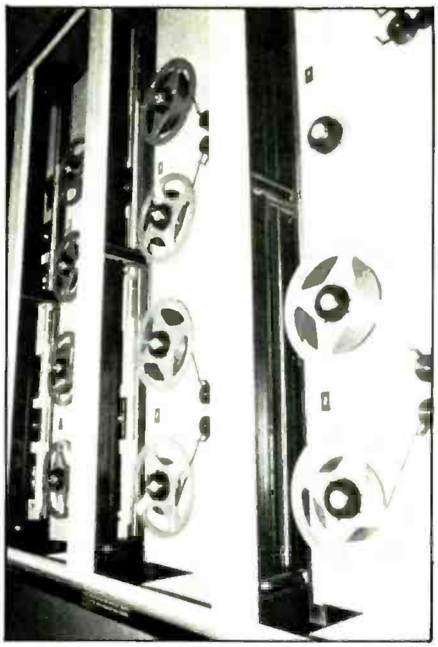
— 35mm and 70mm — in use at EPCOT, along with countless video screens. Over at the Magic Kingdom, which opened 10 years before EPCOT, 16mm film equipment is also used, but during the design of EPCOT the decision was made, in the interest of longevity of the medium and ease of handling, to transfer all of the film shot in 16mm for EPCOT to videotape and disk.

Multitrack, Multimedia, and Synchronization

Dominating much of the floor at EPCOT Central are 36 loop-bin tape decks; there also is space for 12 more. As with all of the equipment at EPCOT, the loop-bin decks were built to specifications of the park's creators, WED (Walter Elias Disney) Enterprises — in this case, by Pacific Recorders, of San Diego, California — and are said to be the first loop-bins built to handle two-inch tape. The tape is stored in a horizontal bin whose size and vacuum pressure are continuously variable to accommodate a wide range of loop lengths — up to 2,100 feet, or 28 minutes at 15 IPS — and also cleans the tape as it passes. Like everything else in the room, the decks are painted orange and gray to match the color scheme of the Sperry mainframe equipment, which make EPCOT Central the first totally color-coordinated audio/video installation this reporter has encountered. (The Loop-bin decks are so unique that they are the only major piece of American-built audio hardware in use at the brand-new Tokyo Disneyland — everything else on the audio side there is made in Japan.)

The loop-bins feed audio and control signals (through 150-ohm balanced lines that are up to two miles long) to the major exhibits, including the Circle-



L.J. Scully background music replay transports with 14-inch reel capacity.

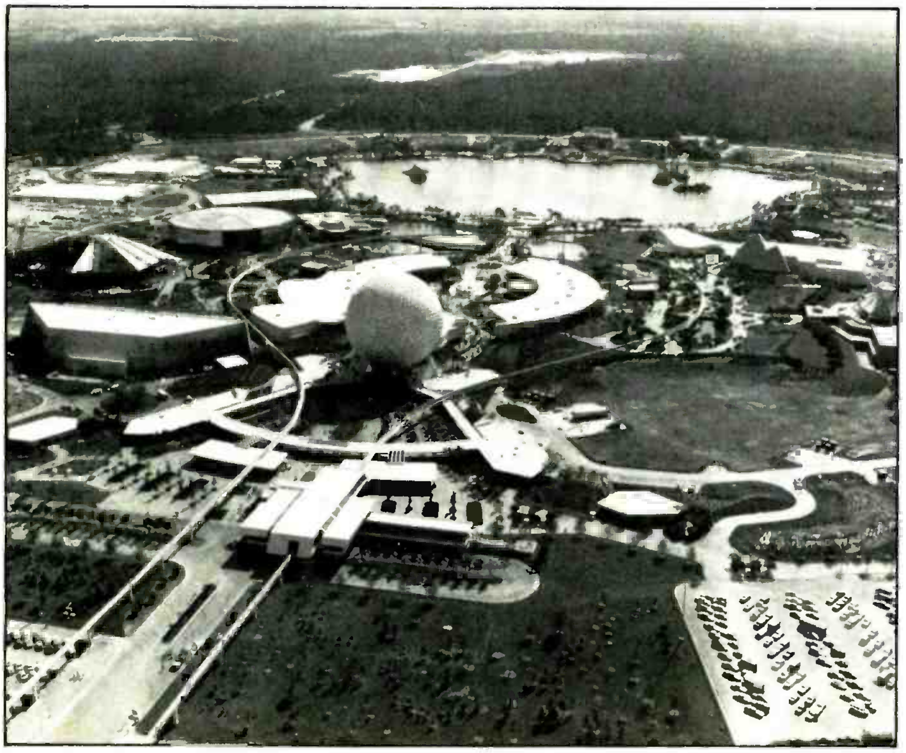
Vision theaters at the Canadian and Chinese pavilions, and the film/audio-animatronics show at American Adventure. The Canadian film is a travelogue, with nine screens set in a circle, each with its own amplifier and speaker, while a 10th speaker hangs from the ceiling. Besides the 10 audio tracks, the tape loops contain foreign-language narration tracks (which we'll get to in a moment), and a modified SMPTE timecode track.

The operation of the theater is simplicity itself: the audience files in, and attendant closes the doors, reads an announcement, and then pushes a button. Back at EPCOT Central, the loop-bin starts up. A computer at the theater, known as a MINIDACS, starts reading the timecode, and its memory unit issues commands that turn off the lights and start the projectors (not all at the same time), while the music swells and the narration begins.

The show at American Adventure is presented on a wide proscenium stage, and uses rear-projection on a scrim, and several other movie and slide projectors. Here the timecode track cues a custom-made hard-disk animation computer built by Data General, which opens and closes a main curtain, flies and moves sets in and out, and handles the incredibly fluid, 24 actions per second, audio-animatronics sequences: figures of Will Rogers, Mark Twain, and other famous and anonymous historical characters speak, move, and do neat tricks like play banjos and twirl lariats.

In the Universe of Energy, audio-animatronic dinosaurs romp through a primeval setting, and the timecode triggers a "smellitzer" device, which helps set the scene by releasing pseudo-sulfurous fumes. In other exhibits, the timecode is fed back to EPCOT Central, and used to trigger interactive videodisc players in a proprietary process that is, apparently, a first.

Everywhere, the sound at EPCOT is excellent. This is due in part to the fact that all of the tapes being used are first-generation analog dubs from digital masters, and also to the quality of the playback equipment in the theaters themselves. All of the amplifiers and speakers were designed, according to WED engineer Dave Spencer, to have as flat a response as possible. "It's very different than a normal movie theater," he says. "There's no Academy roll-off on the audio." Some of the installations use subwoofers to ensure solid bass, which comes in handy for many of the naturalistic sound effects. The signal-processing equipment only controls level, and some equalization; located both at EPCOT Central and in some of the theaters, the processing gear is adjusted infrequently.

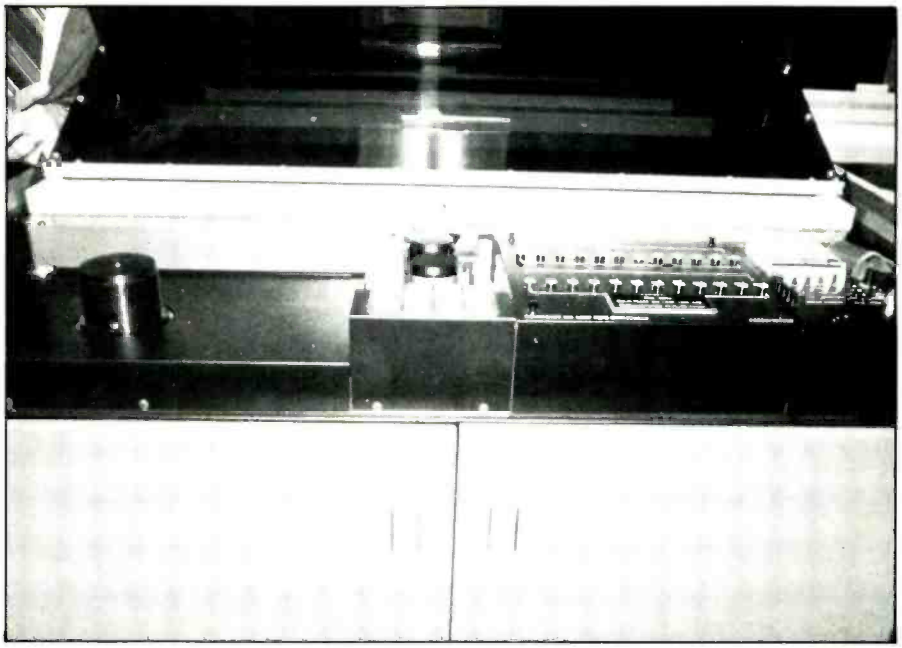


Wireless Systems

The first hint that the EPCOT visitor is in for exposure to some interesting

radio technology comes when you turn off Interstate 4 on to one of the three interchanges that serve the 43-square-mile Walt Disney World. Signs instruct you to turn on the car radio for parking information. All told there are four low-power AM systems in use covering the highway network, and which explain to visitors how to get in and out, and where to park, eat, shop, sleep, and play.

At the EPCOT entrance, hearing-impaired and non-English-speaking visitors can pick up headphone sets known as "Personal Audio Listening Systems." In most of the theaters and rides, frequency-modulated infrared transmitters, mounted in the ceilings, over the curtain, or in the tracks beneath the ride vehicles (the vehicles themselves have built-in re-transmitters), broadcast narration and/or dialog for the show, which is picked up by the headphone set. At the time of writing this article there are only three languages available, each on its own channel, but the system, made for Disney by Sennheiser, is capable of carrying nine, with an audio bandwidth of at least 12 kHz. Extra tracks on the loop-bin tapes and VCRs, or additional audio cartridge decks, supply the source material.



One of 36 two-inch loopbin masters made for EPCOT Center by Pacific Recorders.

The transmission frequencies used in all of the exhibits are identical, so one receiver serves the visitor for the whole day. Selection of the receiving frequency (and the language) is made when the visitor gets their receiver — instead of there being different receivers for the different channels, the tuning elements are in the rechargeable battery packs. All the attendant has to do is select the right battery, and snap it in.

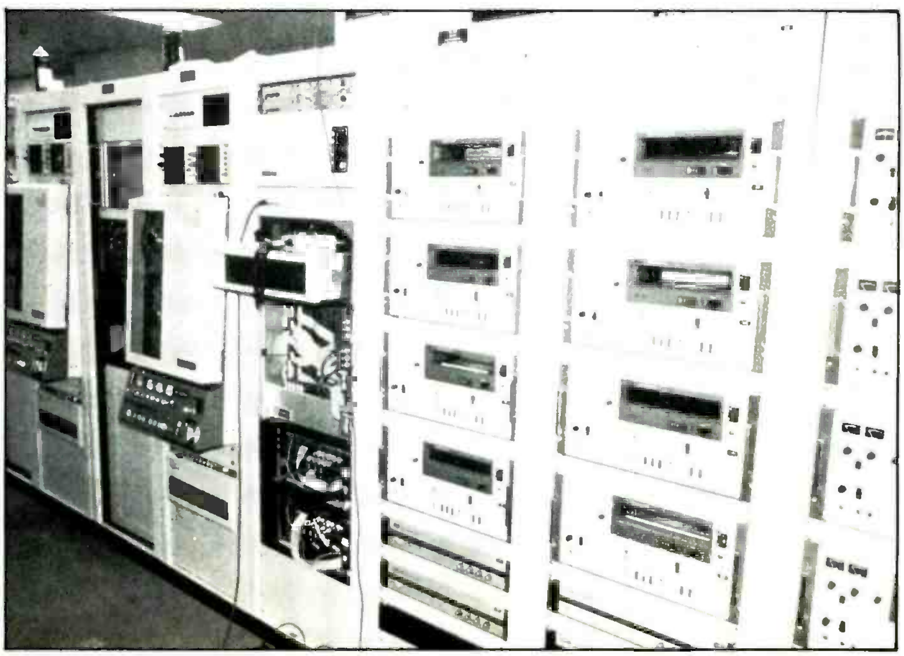
A complex two-way wireless system is used for parades through World Showcase, which originally were scheduled for twice daily, but have had to be cut back drastically due to the unexpectedly large crowds drawn by the park. Each parade float carries an FM receiver, along with an audio amplifier and speaker system. On top of the American Adventure building is an FM transmitter with 11 separate wide-spectrum audio channels, which are fed from an Ampex MM-1200 24-track at EPCOT Central.

The receiver on each float is locked on to one channel, which it plays continuously throughout the parade. Each float also carries a low-powered FM *transmitter* (operating on a different band) that continuously broadcasts an identification code, based on an ASCII character set. This signal is received by antennas embedded in the concrete beneath the parade route, and serves to relay to the central computer the float's location. The computer fades out the background music (coming from the L.J. Scully decks) in the permanent speakers in the immediate vicinity of the float, and ramps up a different audio signal — the same track of the multi-track deck that is feeding the float itself. The result is that parade watchers hear only one audio track at a time, coming both from the float, *and* the permanent speakers.

Maintenance and Operation

With all of the film, video, audio, and switching equipment, connected by hundreds of impossibly long cable pairs, EPCOT could be a maintenance engineer's nightmare. If something goes wrong during a major show, hundreds of people who may have waited in line

Below: A bank of interactive videodisk players in the central equipment room. **Right**: One-inch and shuttling ¾-inch video transports.



for over an hour to see it could be very disappointed. To keep problems at a minimum, there are at least 11 engineers stationed at EPCOT Central during each shift to monitor trouble calls, and another 10 or so in the field, ready to jump at a moment's notice.

Less-technical personnel operate and monitor the individual attractions, but they too are trained to respond quickly to trouble. An employee at each theater monitors all of the operations of the show on a small video screen; if a malfunction occurs, he or she notifies the engineers at Central. The operators decide whether the problem is serious enough to cancel the show, or whether it can be fixed while the show is running, or between shows.

For example, a blank rear screen at the Canadian theater, where the primary audience focus is towards the front, would not be cause for stopping the show, although the crew would take care of the problem before starting the next one. On the other hand, at the China pavilion, where the audience constantly changes its focus, a burned-out projection bulb might be considered grounds for aborting the presentation. Says EPCOT project manager Jerry Aldrich, "We'd rather give the guests no show at all than a *bad* one."

Overview of EPCOT Central with two-inch loopbin desks and control equipment.

When trouble occurs, a radio call goes out to one of the field maintenance teams, each of which is responsible for a small area of the park, and is no more than five minutes away. The quickest response time is at the China pavilion —the projection shop is located right in that building.

A more sophisticated trouble-monitoring system should be on-line any time now. The individual video monitors in each theater will be wired back to EPCOT Central, where an operator can call up the data from each theater on his own touch-sensitive video screen. At the same time, the audio output, as it appears at the speakers in the theater, will be fed back to Central and switched to the operator's headphones so that he can do "real-world" confirmation of signal presence, level, and purity. The new system will allow Central operators to reach "go no-go" decisions faster, and will speed up maintenance-team response time.

Unforeseen Problems

Despite all the painstaking technical planning and development, there was one factor that was not wholly taken into account during EPCOT's design: the hugh number of people (referred to as "guest acceptance" in Disney parlance) who have flocked here since its opening in October 1982. Last December 28, for example, paid admission to all of Walt Disney World was 123,000 — 50% higher than the figure the "Official Guide" to the place describes as "body to body."

Among other things, such popularity has forced a re-arrangement of maintenance schedules. "Originally, Future World was to be open from 9 a.m. to 6 p.m. The day shift would clean projectors and film at World Showcase, while the second shift could maintain the equipment at Central, and clean the projectors at Futureworld. But due to the demand, we have to keep both areas open all the time, so we've had to reroute and reschedule the staff, and do all of the maintenance work after midnight."

Another problem is that the ebb and flow of absorbent human bodies causes sound levels to constantly shift, particularly in the outdoor areas. In the theaters, sound levels are carefully maintained by WED engineers who come to EPCOT once a month and test each space with an SPL meter. The outside areas, however, although they are kept at constant *electrical* levels, are trickier.

"A common complaint we get from guests," Aldrich offers, "is that everything is so *loud*. It may be too loud in one spot at the beginning of the day, until 600 people crowd in, and then it's fine. We tried some automatic level adjustment at Magic Kingdom, but it never worked. We used Altec and TOA automatic mixers, but we would get unpredictable level drops. It may have had something to do with the sound changing as it went through the transformers in the distribution system, or the horses and trolleys on Main Street affected the levels.

"It would be easier at EPCOT, because we could program the computers to adjust the levels according to the time of day. Of course, then you're gambling that the crowd will flow the same way and peak at the same time every day. In reality, they just seem to stay all the time. The other alternative would be to put feedback equipment at the computer or the output end."

A Changing Environment

Although EPCOT is far from the "Prototype Community" that Walt Disney envisioned for this Florida land

SOUND PRODUCTION FOR EPCOT

Although glimpses of the technology for presenting audio and video material at EPCOT are available to the paying customers, the artistic and technical forces that went into the creation of the program material are far less visible. "People who know film see the Canadian theater and say, 'It's just a two-reeler,' " offers Nelson Meacham of Walt Disney Studios. "What they forget is that there are nine screens, so it's actually an *18-reeler*."

Producing the multimedia programs for EPCOT was a collaborative effort between two arms of the Disney empire: Walt Disney Studios of Burbank, California, the filmmaking operation whose movie accomplishments have ranged from *Steamboat Willie* to *TRON;* and WED Enterprises of Glendale, CA, which has primary responsibility for the theme parks and their equipment. The complexity of the project was a challenge even to the formidable resources of both companies.

Meacham, who is a supervising engineer in the studio's sound department, and Shawn Murphy, supervising mixer on the EPCOT film projects, explain how they met the challenge. "We had no dubbing theater with nine screens," Meacham recalls, "so we built a scaled-down version of a Circle-Vision theater with nine video projection screens, each one perforated so a speaker could be placed behind it. [Another speaker, for narration tracks, was hung overhead.] We didn't want to fool ourselves by trying to do it in a recording studio — we wanted to simulate the real thing as much as possible."

The films were transferred to Betamax videocassettes, with timecode, of course, and played back on the new dubbing stage with nine Sony Cinemascope video projectors. "WED developed a code format for dubbing and scoring," says Murphy. "It was basically non-drop SMPTE, but with a footage counter — running at either 24 or 30 frames-per-second — in the user-bit positions. The zero mark for the footage corresponded with the one-hour mark on the SMPTE [timecode]. That's so when we backed up past the start point, the SMPTE readers wouldn't be looking at 'before-midnight' numbers, and get lost." The generators and readers utilized during the project were built by Gray Engineering Laboratories.

The recordings used on the final film tracks came from many sources. Some sound was recorded in the field while the cameras were rolling, on mono or stereo Nagras, but not much. More effects were recorded after the fact on location by Disney engineers, while others came from the studio library, were recorded on the Disney or Goldwyn studio Foley stages, or were generated synthetically. "Ninety-five percent of the effects were recorded analog," says Murphy. "They were copied with Dolby, and then cut, before being transferred to the digital dubbing masters." On the other hand, 90%

. . . continued overleaf —

before his death — after all, except for the guests in the 14 hotels, villas, and campgrounds at the site, nobody actually *lives* here — it is a new concept in entertainment; a kind of ongoing Worlds' Fair, designed primarily for entertainment, and secondarily for education, whose operation is almost totally automated.

There's a lot of corporate advertising going on, of course (Kodak, General Motors, American Express, Kraft, and Sperry are just some of the companies who sponsored and collaborated on the exhibits), but it is all filtered through the minds and design concepts of Disney's own "Imagineers."

More importantly, it is not static — new attractions will open old ones will be replaced or refurbished while the technology of the presentations will get further and further out. Omnisphere, for example, a ride scheduled to open in the Fall of 1983, will use a 4,000-line video projector to show images from the Landsat satellite on the inside of a giant dome. At the Equatorial Africa pavilion, still in the planning stage, laser graphics and film projection will be combined under computer control.

The technology — audio, video, and computer — already in place is designed to handle all of the needs of the center, both present and future. It should be interesting to check back in a few years to see how it's holding up. ■■■

SOUND PRODUCTION FOR EPCOT

. . . continued —

of the music tracks (and all of the narration) was recorded directly on 3M Digital Mastering equipment.

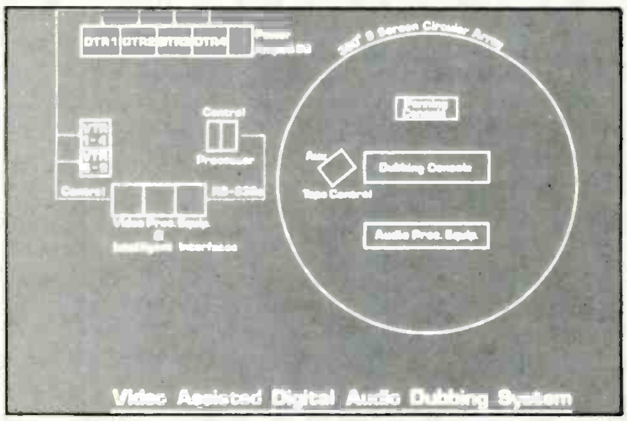
Digital Recording and Editing

Sometimes, the digital recording stages involved some effort. For music at the American Adventure, for example, the Philadelphia Orchestra was recorded in its home, the Academy of Music, on a 3M digital multitrack belonging to, and brought across the country by, Disney. Supervising the sessions was Jack Renner of Telarc Records, who specialize in digital symphonic recordings. For other exhibits, the Royal Philharmonic and the National Philharmonic of London were recorded by Eric Tomlinson at EMI Abbey Road Studios, using at various times a 3M machine belonging to the studio, and one of Disney's. All remixing of the various digital multitrack tapes was done in Burbank.

The number of tracks that had to be assembled for the master tapes was large. "Each screen of a nine-panel show has music, dialog, ADR, guide tracks, hard effects, and Foley effects," says Meacham. "Sometimes several units of each." The tracks were preassembled on 32-track digital tape, which took two or three passes before their number was reduced to a manageable level. In the dubbing studio were located three 3M 32-track machines, and one four-track deck. "The four-track was a transfer machine for slipping tracks," Murphy recalls. "We know we were going to have to adjust things, and we didn't want to have to worry about generation noise."

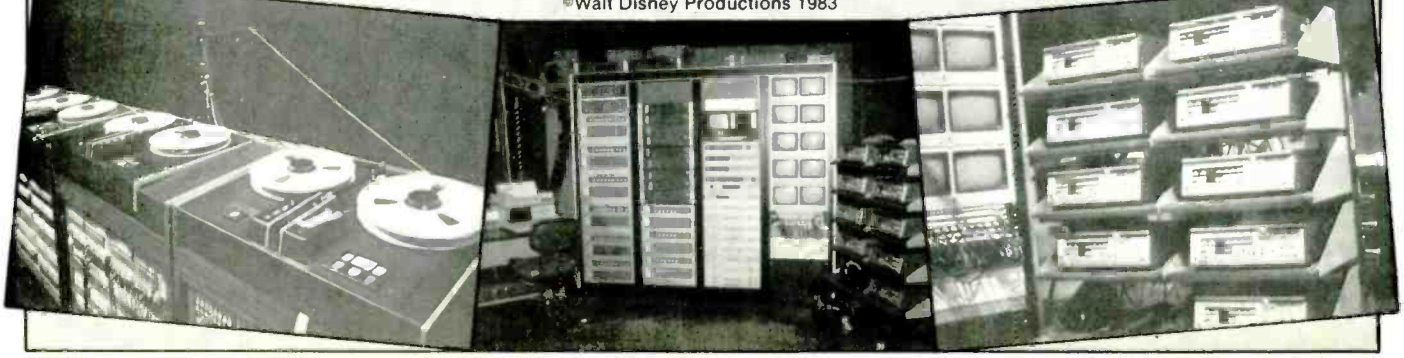
The four-track digital machine had another use, early in the process of putting together the audio tracks. For some of the

Above from left: Walt Disney Studio's central mixing room, equipped with a Harrison TV-3 console and 360-degree, multichannel monitoring; three of the nine video projection screens; narration monitor speakers mounted above the console area.



©Walt Disney Productions 1983

Below from left: Three 3M 32-track Digital Mastering Systems for dubbing multichannel music, dialog, ADR, Foley, and effects tracks; central electronics, audio/video synchronization, and monitor racks; mulitiple videocassette transports.



less-complex exhibits, audio from the 3M digital was transferred to a Soundstream digital recorder, using a data interface convertor developed by Soundstream for the digital-to-digital transfer process.

"Editing on the 3M is time consuming," says WED's Dave Spencer. "We wanted to be able to put, for example, 17 loops of a one-minute program on a single cartridge. The 3M system would have had to do that in real time, while Soundstream's computer [a complex digital editing system, Instant Access, based at the company's Hollywood location] could just spit the data out."

After the editing, the finished loop was transferred back to the 3M digital format to make a printing master — "We wanted to keep all the masters in the same format," says Spencer. The Soundstream-designed computer could handle up to eight digital tracks, but wasn't called upon to deal with more than four at a time.

Post-Production and Mixing

Keeping the nine video and four audio machines on the dubbing stage in sync was the job of a rack of BTX Shadow synchronization equipment. A Hewlett-Packard 9826 computer was used, in turn, to control the Shadows. "Since they're software-oriented," says Meacham, "The BTX units could be addressed rapidly by a computer. With standard SMPTE you could get everything to start together, but with the computer and our code, each synchronizer could be addressed individually, in either SMPTE or footage numbers, and the computer could handle offsets for each unit."

Special considerations had to be made for the dubbing console as well. Disney Studio's 40-input Harrison TV-3 was modified so that its eight stereo subgroups could be used as additional monitor channels. "The console has 24 bus outputs," says Murphy, "which should be enough for a 10-channel master, but there weren't enough faders." The TV-3 board also was equipped with direct outputs, to free up more faders, and was rewired so that it could simultaneously record — on the same piece of 32-track digital tape —music-and effects-only mixes. This provision enabled the engineers to make last-minute changes in any part of the presentation, without disturbing the other mixes.

The Harrison TV-3 proved adequate for the pre-dubs, but for the master mixes a second console had to be rented and brought in. Depending on the project, this was either another Harrison, an Auditronics, or a Sphere, with between 40 and 48 inputs.

"The sound effects came up on the rental console," explains Murphy, "while we used the main console for music, narration, and dialog."

None of the consoles used automation: two pairs of hands were required for the pre-dub stage, while the final mixes took three engineers working together. Besides Murphy, also working on the mixes were Andy Bass and Richard Portman.

The final 32-track digital master was dubbed, track for track, on to two-inch analog tape at 15 IPS with no noise reduction for use in the loop-bin decks at EPCOT, using a 24-track Ampex MM-1200 deck. While there are only 10 main audio tracks for each show, "The analog tapes are pretty full," according to Murphy. There are additional tracks for announcements, pre-show music, foreign-language narration, and timecode to cue the computers that control EPCOT's projection and animation systems. The code, which is inserted at the final digital stage, is yet another custom SMPTE modificaiton: since the show computers are designed to read the clock and update themselves 24 times each second, the code counts 24 frames-per-second instead of 30 FPS.

During the mixing stages no attempt was made to compensate for room acoustics at the EPCOT pavilions. Instead, the first analog dubs were taken to Florida and played back in the theaters through a portable filter set. An engineer would dial in whatever equalization was needed, and call the numbers back to California. Then, a matching set of filters was hooked up, the numbers dialled in, and the digital tapes redubbed to analog, through the filters. "The amount of compensation was minimal in all of the theaters," says Meacham, "except China — there, the tape matched the room exactly." ■■■

CIRCUIT DESIGN AND EVALUATION

THE COMPUTER AS A CIRCUIT DESIGN TOOL ... OR ADVENTURES IN FANTASYLAND

Detailing the Multiple Applications of COMTRAN Program Software Configured to Run on Hewlett-Packard Desk-Top Computers

by Peter Butt

Computers have always held a fascination for those who consider themselves technically inclined. Speaking personally, the advent of the minicomputer caught my interest in a voyeuristic sort of way from the onset of my awareness of such a device circa 1977. A fairly brief examination of the periodicals dealing with the minis of that time did not show any indication of early delivery on the promise of affordable numerical manipulative power in a convenient package. The majority of the micros of that day were bread-board enterprises undertaken by those seekers of knowledge blessed with sufficient time to work out the bugs in both hardware and software, so that the persnickity creatures would work in some way that could be interpreted as serving a kind of useful purpose. The microprocessor of those days was substantially an end in itself, as far as the owner/tinkerer was concerned.

Projects of this kind have the air of what might be called The Time-and-Money-Sink Syndrome. This writer had confronted similar situations during those years past when the installation of outlandishly huge engines in automobile bodies of indefinite vintage was one of his vices. What was once an infatuation has progressed, alas, to affliction and, anon, to phobia. The memory of the passage of that honeymoon remains with me even now. As in the case of the reformed profligate, the memories of diseases, past, serve to inhibit adventures, present. Although, lookin' never hurt nobody, you understand.

My awareness of the minicomputer remained little more than that over the intervening years when my attentions were directed more toward hand-to-mouth concerns leading to near-term gratifications (i.e.: survival).

That has been the case ... until fairly recently.

Earlier this year my infrequent reconnoiters through the local computer stores brought me to the realization that the mini has been growing to be a tool whose time may indeed be upon us. The maturity of the Tandy TRS-80, the DEC PDP-11, LSI-11, the Apple IIe and Lisa — machines that now have the capability of performing really useful tasks beyond solipsistic video games. Then, of course, when IBM jumps into the ring, you know something's up!

Well, folks, *could* be.

Before we are carried away by the passion of the moment and rush headlong into the arms of our friendly, neighborhood computer vendor attempting to force him/her into submission with fistfuls of cash, it is well to remember that the hardware is rarely the whole package. As in the case of the Hot-Rod Analogy outlined above, the practical utility of the machinery is a very real consideration for the pragmatically oriented. Eleven-to-one compression ratios are less exciting when the volatility of predominantly available motor fuel resembles that of warmed-over cherries jubilee. So, too, computers are less appealing when the available software directed towards electronic engineering applications are more specific rather than general; sparse rather than plentiful.

Cherche le Software

Inspection of the listings of available software for the Apple II+, IIe, III machines[1] reveals nothing for engineering applications, and only a few utility programs for minor math and statistics functions, plus a couple of plotting and graphics routines. The collected listing of TRS-80 programs[2] shows about a dozen programs oriented toward electronic analysis and design problem-solving.

These programs appear to be rather disjointed, likely requiring that programs be loaded, run, results stored, any subsequently required program be loaded, stored data be re-entered and run, results stored, etc., *ad nauseum*. If you don't want to do a resistive attenuator or a third-order Chebychev in a particular topology, with active elements of anonymous choice, then you've come to the wrong place. Program repertoire for the DEC machines doesn't appear to be readily available to the vulgar and, as of this writing, IBM PC programs seem to be primarily business and game applications.

The nature of the programs described in the TRS-80 catalog appear to be similar in function to many of the Hewlett-Packard Users' Library of programs for the company's HP-67, HP-97, and HP-41 hand-held programmable calculators[3]. As a bigotted HP-67 owner and user for several years, it is hard to generate enthusiasm for computation systems that don't seem to clearly offer a great deal more than the romance of a CRT display, a printer, and perhaps a slight speed advantage over my friendly mental prosthetic for half a dogs' age.*

This state of affairs is enough to cool one's ardor. Still ... they do look nice.

What is required for the efficient doing of useful electronic circuit analysis and design is an integrated package of capabilities that do not restrict the user to specific circuit topologies, such as ladder networks, attenuator designs, active and passive filter networks, or other narrowly-defined computational chores that might fall into one's daily design routine ... or not ... as the case may be. Such a program would lend itself to more general application, by approaching circuit modeling from a nodal viewpoint that allows for flexibility in selection of components, and introduction of current and/or voltage sources controllable from assignable nodes.

IBM offered a program matching this description over 10 (15?) years ago. It was called "Electronic Circuit Analysis Program" — ECAP for short — ran in Fortran IV, and was not "user-friendly" to any overwhelming extent. (Perhaps "user-aggressive" might be a more apt term.) I used this program for a time during involvement in the military-industrial complex in the days of my misspent youth. If memory serves, the ECAP experience was an awkward one. It was run on a time-shared multi-user system, and took hours to complete what are now fairly elementary circuit analysis problems of only two or three stages.

It turns out that my old friend, ECAP, has been transcended by an enterprising soul by the name of Deane Jensen (of transformer fame). Jensen has written, and continues to write, a program, known as COMTRAN, which does everything ECAP does and, as they sometimes say on TV, much, much more[4]. COMTRAN is the blanket name

*It should be pointed out, however, that those of us who already own an Apple, Atari, Commodore 64, VIC 20, Radio Shack TSR-80, IBM PC, and similar micros capable of running BASIC and similar "user-friendly" languages, can write a certain amount of their own, customized engineering software. Ethan Winer's recent series of articles — in particular his Applesoft BASIC listings for calculating op-amp resistor networks, published in the April 1983 issue of *R-e/p*, and pad and attenuator circuits, to be published in the December issue — contain useful, easily modified programs for the inveterate circuit designer and builder — *Editor*.

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for a number of analysis programs and utilities that are loaded into RAM, and accessed as required by the user.

Basic "kernel" of the software cluster is the AC Circuit Analysis Program. AC-CAP allows construction of a circuit model in a nodal configuration consisting of resistors, capacitors, inductors, and voltage-controlled current sources. Input and output nodes are selectable, with node "0" being permanently assigned as reference, or "ground." The "hitch" is that COMTRAN is configured to run on the Hewlett-Packard desk-top computers — 9845B/C, 9816S, 9826, 9836, and 9000 frames — which are the "heavy iron" compared with the denisons of ye neighborhoode computer store. Minimum 16-bit RAM required to run COMTRAN on a 9845B is 187 kilobytes. Casual perusal of your HP catalog will apprise you that the cost of these systems starts at around $10,500, and proceeds upward from there. If you choose the 9816S, it also might be nice to have the low-cost 82905B printer, and a cute little plotter such as the 7470A. Of course, a mass storage device such as the 9121A flexible disc drive would be a necessity.

It's only money.

This may sound like the gold-plated Rolls-Royce approach to the mainframe problem, I will readily admit. The fact is though, that to be able to cope with electronic-circuit simulation problems of any really useful complexity, while utilizing the sophisticated graphics output features and data matrix operations capability that facile program manipulation requires, coupled with the human-factor need for tangible results in a fairly short computation time, there is *no* other real choice.

If one is in this game for the sake of one's beans & hot dog ration, then one needs to understand that productivity and cash flow rest on the quality and reliability of one's tools. It did not take many forays to the local computer stores to establish that Hewlett-Packard, as my parochial instincts suggested, is the *only* game in town as far as engineering applications of computers are concerned. A 16-bit frame is *de rigeur* for all HP machines. A 750 kilobyte RAM complement installed in the 9816S allows processing of networks having up to 95 nodes and a maximum of 400 components. This magnitude of computing hardware just isn't out there in the Personal Computer market of 1983.

But, Dear Reader, let us not be morbid.

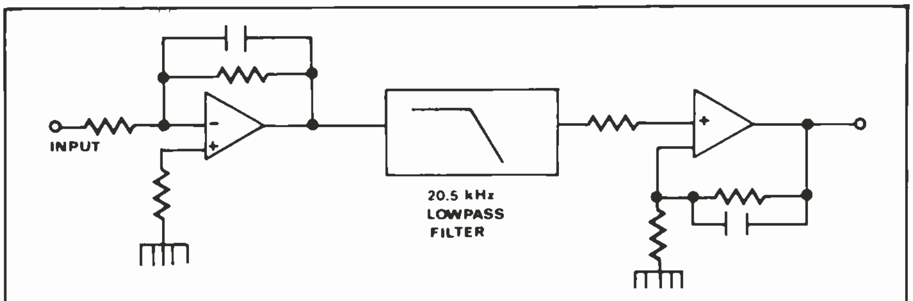


Figure 1A: System Block Diagram.

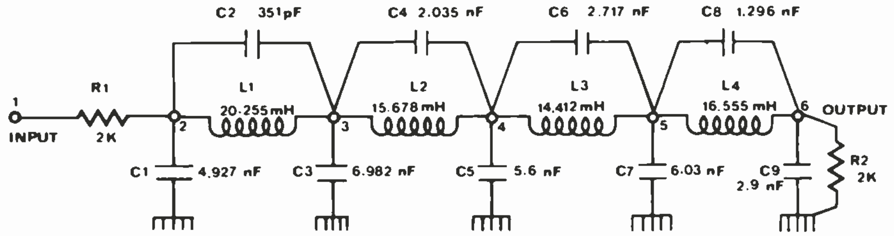
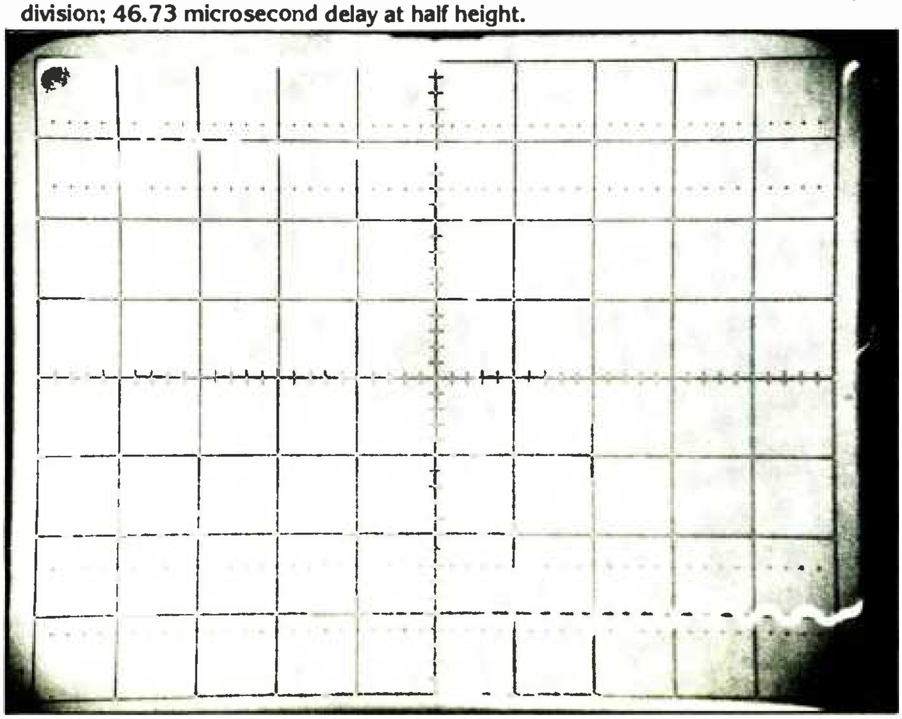


Figure 1B: Anti-aliasing Filter detail showing scaled component values for a 20.5 kHz passband.

Figure 2: Real filter 1 kHz square-wave response. Larger trace is the filter input; wavy trace the filter output. Vertical scale: normalized. Horizontal scale: 0.1 millisecond per division; 46.73 microsecond delay at half height.



AC Circuit Analysis Program

There's more to COMTRAN than AC-CAP, but we'll get to that later. Briefly, AC-CAP enables the user to enter, via the computer keyboard, a circuit consisting of passive circuit components, circuit junction nodes, and voltage-controlled current sources. Any circuit can be entered this way or loaded from, or printed to, mass storage; it also can be edited with infinite flexibility after entry by either means. Component values may be either positive or negative, non-zero values. Active or passive components can be added or deleted or modified at any time during the modeling process. Input and output nodes can be selected or edited at any time during the modeling process, and the results of the circuit response with respect to magnitude, phase, phase delay, group delay, and complex impedance versus frequency can be determined and displayed at any time as a rectilinear two-dimensional graphic CRT display, or as a CRT or printer tabulation. Frequency points displayed can be selected in either linear or logarithmic intervals. Graph vertical scales invariably are shown as linear decibels for magnitude plots; in linear electrical degrees for phase graphs; and in linear seconds for phase delay and group delay plots. All AC-CAP data displays are in the frequency domain; it is possible to choose scales and scale graduations for each of the six frequency-domain plots. If the effects of component changes or nodal changes are to be observed comparatively, the necessary edits can be made without destroying traces previously run — this permits the effect of such changes to be observed progressively, overlaid on the same grid.

Not only can the center values of components be specified as well as edited, the tolerance of that value may be specified on a component-by-component basis. AC-CAP's Tolerance Mode function permits graphic display

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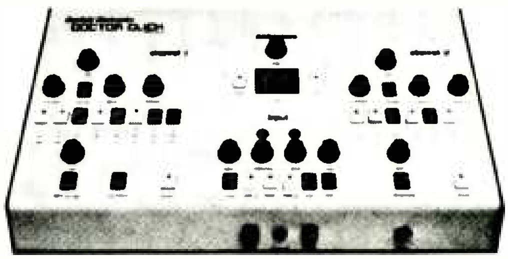

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The brand to brand problems of timebase, voltage level and polarity are solved by the **Doctor Click's** diverse output capability.

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

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

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

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

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

The ability of the **Doctor Click** to accept external clocking or either of the types of FSK sync to tape codes allows sequencers, drum machines and synthesizers to be synced to any existing track.

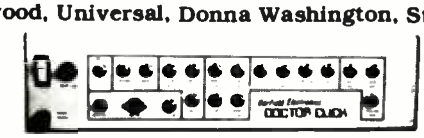
The pulse shaper circuit turns a pulse from an instrument into a trigger waveform allowing synthesizers to sync to a drum fill.

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

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

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

Used on tracks by **Brian Banks, Tony Basil, John Berkman, Michael Boddicker, Kim Carnes, Suzanne Ciani, Joe Conlan, Chris Cross, Bill Cuomo, Jim Cypherd, Paul Delph, Barry DeVorzon, Don Felder, Paul Fox, Dominic Frontier, Terry Fryer, Albhy Galuten, Lou Garisto, Herbie Hancock, Johnny Harris, Hawk, James Horner, Thelma Houston, Michael Jackson, Quincy Jones, Jeffrey Kawalek, Gordon Lightfoot, Jerry Liliedahl, Johnny Mandel, Manhattan Transfer, Paul Marcus, Jason Miles, NBC Movie of the Week, Randy Newman, Keith Olsen, Paramount, Joel Peskin, Oscar Peterson, Greg Phillingaines, Jean-Luc Ponte, Steve Porcaro, Phil Ramone, Lee Ritenour, Steve Schaeffer, Mike Sembello, Mark Shifman, John Steinhoff, Sound Arts, Ian Underwood, Universal, Donna Washington, Stevie Winwood, Pia Zadora.**



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PATENT PENDING

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of the effects of random tolerance variations on the circuit performance. The results of tolerance variations on the circuit under evaluation then are plotted as a dual trace, showing the high and low limits of circuit performance on the appropriate frequency domain graph previously selected by the user. The value of this feature is obvious to those who must wrestle with component budgets and need to isolate the critical tolerance parts, as well as to determine the limits that tolerances can be allowed to have and still maintain specified circuit performance.

The AC-CAP Impedance Mode function permits determination of the magnitude of the real and imaginary components of the complex impedance appearing at a single node of choice to be plotted on a graph versus frequency, or in tabular form on the CRT or printer showing magnitude and phase angle versus frequency. Any node may be selected for impedance determination, including the specified input and output nodes. Only one impedance node may selected at any given time, and the impedance is always indicated with respect to the ground, node 0.

Square-Wave Response

Once a circuit appears to be performing as required in the frequency domain, it is usually of interest to take a look at it on the ol' scope, and see how a square wave looks. COMTRAN has this capability, too. It's done by generating the circuit transfer function in the form of amplitude and phase response data over a range of frequencies that are 128, 256, or 512 multiples of the reciprocal of the time window of interest. For a 1 kHz square wave, we would choose a time window of 1 millisecond, since such a period implies a frequency of 1 kHz as its lowest frequency component. The 128, 256, or 512 points of magnitude and phase response data, starting at the base frequency, are stored as input for inverse Fourier transformation, and displayed as time-domain data graphic plots by the companion to the AC-CAP program called S-WAVE: more about that later.

Perhaps the COMTRAN program's most novel capability is its Optimization Mode. Given optimization-objective data in the form of discrete points or band-defined information in the frequency domain, a circuit can be optimized, through the variation of specific components selected by the user, until the target data is matched by the circuit response. This is an incredibly powerful capability to have, since it permits the user to synthesize a system having desired magnitude, phase, and impedance response; such a task would be mind-numbing if a large number of components are involved, and raw brain power is the only tool at hand.

As the optimization process proceeds, the program displays its progress as each iteration of the optimization is completed. Latest values of the optimized components are displayed, along with a number for each called "Gradient," which shows the relative sensitivity of that particular component with respect to its effect on the system parameters being optimized. Such information is useful in spotting components that have greater or lesser impact on the desired circuit performance. Appropriate component tolerances then can be specified, and their acceptability verified with a Tolerance run.

CIRCUIT DESIGN EXAMPLE: Digital Anti-Aliasing Filter

The major point to be illustrated here is that the capabilities of this computer-based system are often boggling. By way of an example of AC-CAP's potential, a recent project concerned the generation of an elliptical anti-aliasing filter similar to the one shown in Figures

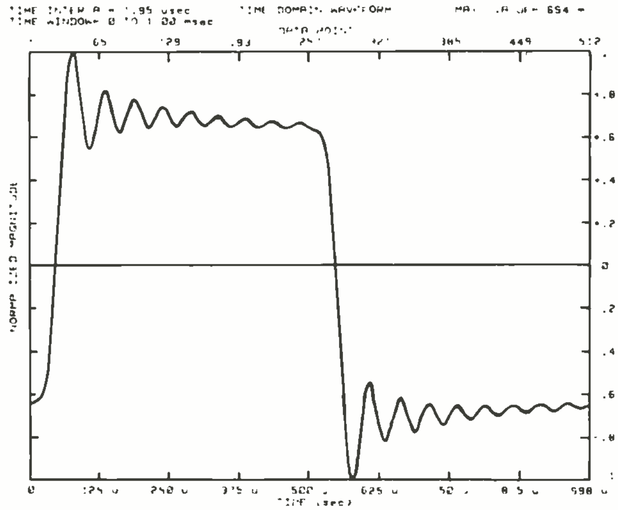


Figure 3: Optimized Model Filter 1 kHz square-wave response after 24 optimizations.

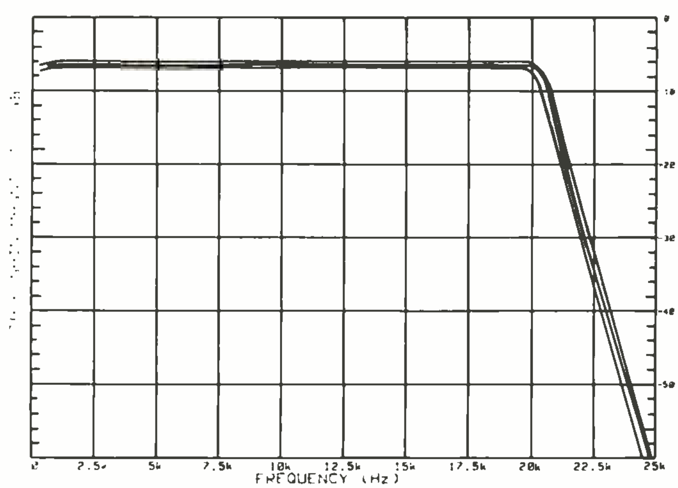


Figure 5: Magnitude Response of optimized filter after 24 optimizations; +2% component value tolerances.

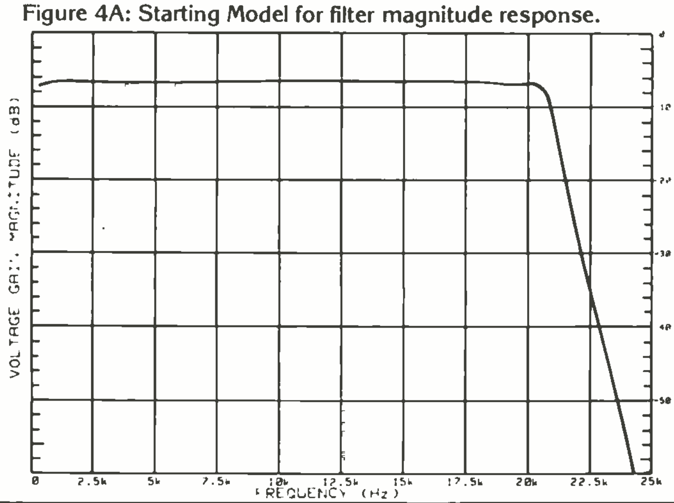


Figure 4A: Starting Model for filter magnitude response.

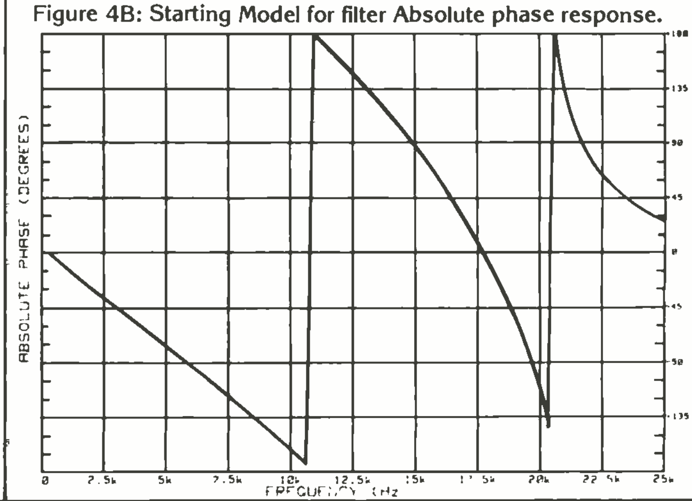


Figure 4B: Starting Model for filter Absolute phase response.

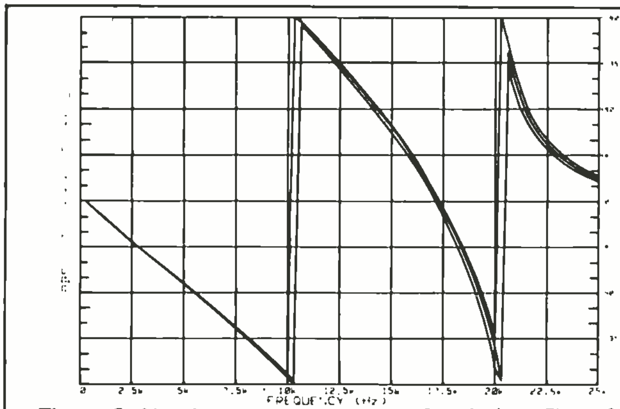


Figure 6: Absolute Phase response of optimized filter after 24 iterations; ±2% component tolerance limits.

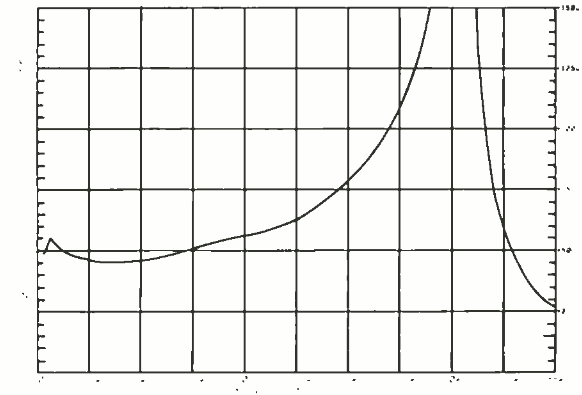


Figure 7: Absolute Group Delay of optimized model filter (24 iterations).

1A and 1B. The aware will recognize this block as typical analog interface system for a digital audio reproducer output stage. The problem motivating such scrutiny is the fact that, as originally figured, the audio signal produced by this system fell considerably short of the startling impact we've all been led to expect from digital technology.

The nature of the problem-solving process requires, that first, the problem be defined. The key element is the anti-aliasing filter which, in this case, has been executed in a thick-film hybrid design. Specifications for the filter exist only as amplitude tolerance-band versus frequency data. Direct empirical amplitude and phase response measurements had to be made and recorded manually to arrive at a more detailed filter performance characteristic. A multipole elliptic filter whose circuit file happens to be included as a demonstration item on the COMTRAN program disk looked like it roughly approximated the observed actual filter response. Very roughly.

The sample filter circuit was scaled in proportion to the ratio of the sample-filter's bandwidth to that of the real filter. All capacitors and inductors were multiplied by a constant factor of 0.9738 as a starting point for the optimization program to begin. (This factor being based on the ratio between the model filter's bandwidth at 7.9 dB below the average pass-band plateau at the upper band edge, and the same relative response frequency for the empirical case.)

TABLE 1A: MAGNITUDE AND ABSOLUTE PHASE DATA OF FIRST APPROXIMATION FILTER AND THE REAL FILTER TARGET DATA.

Voltage Gain Magnitude:

Frequency	Actual (dB)	Target (dB)	Error (dB)
1.00 kHz	-6.71 dB	-6.80 dB	92.10 mdB
1.60 kHz	-6.66 dB	-6.75 dB	88.68 mdB
2.00 kHz	-6.66 dB	-6.75 dB	89.33 mdB
2.50 kHz	-6.67 dB	-6.70 dB	29.20 mdB
3.15 kHz	-6.69 dB	-6.70 dB	10.68 mdB
4.00 kHz	-6.71 dB	-6.70 dB	-8.65 mdB
5.00 kHz	-6.71 dB	-6.65 dB	-60.76 mdB
6.30 kHz	-6.67 dB	-6.60 dB	-73.60 mdB
7.50 kHz	-6.62 dB	-6.60 dB	-16.32 mdB
8.00 kHz	-6.59 dB	-6.60 dB	5.91 mdB
9.00 kHz	-6.57 dB	-6.60 dB	33.83 mdB
10.00 kHz	-6.57 dB	-6.60 dB	31.00 mdB
12.50 kHz	-6.63 dB	-6.60 dB	-33.87 mdB
15.00 kHz	-6.59 dB	-6.60 dB	13.35 mdB
17.00 kHz	-6.59 dB	-6.70 dB	109.87 mdB
18.00 kHz	-6.61 dB	-6.60 dB	-9.95 mdB
19.00 kHz	-6.85 dB	-6.55 dB	-302.43 mdB
20.00 kHz	-6.94 dB	-6.60 dB	-336.86 mdB
21.00 kHz	-12.13 dB	-14.50 dB	2.37 dB

Voltage Gain Phase:

Frequency	Actual (deg)	Target (deg)	Error (deg)
1.00 kHz	-12.98 deg	-12.75 deg	-226.48 mdeg
1.60 kHz	-23.08 deg	-23.60 deg	517.63 mdeg
2.00 kHz	-29.55 deg	-30.60 deg	1.05 deg
2.50 kHz	-37.49 deg	-38.90 deg	1.41 deg
3.15 kHz	-47.67 deg	-49.80 deg	2.13 deg
4.00 kHz	-60.92 deg	-64.60 deg	3.68 deg
5.00 kHz	-76.57 deg	-80.50 deg	3.93 deg
6.30 kHz	-97.40 deg	-102.90 deg	5.50 deg
7.50 kHz	-117.46 deg	-124.80 deg	7.34 deg
8.00 kHz	-126.12 deg	-134.40 deg	8.28 deg
9.00 kHz	-144.00 deg	-153.40 deg	9.40 deg
10.00 kHz	-162.64 deg	-173.20 deg	10.56 deg
12.50 kHz	147.20 deg	134.60 deg	12.60 deg
15.00 kHz	87.76 deg	71.50 deg	16.26 deg
17.00 kHz	27.60 deg	9.50 deg	18.10 deg
18.00 kHz	-9.99 deg	-28.40 deg	18.41 deg
19.00 kHz	-55.92 deg	-75.80 deg	19.88 deg
20.00 kHz	-115.47 deg	-145.00 deg	29.53 deg
21.00 kHz	135.03 deg	114.80 deg	20.23 deg

The magnitude and phase versus frequency data for the empirical filter are given in Table 1A, along with the data for the scaled first-approximation model filter. The -7.9 dB point was selected because the response point happened to fall at exactly 21.000 kHz for the empirical case, and provided a convenient point for a comparative band-edge reference. The observed filter data was used as target magnitude and response data for optimization of the initial trial filter between 1 kHz and 21 kHz. Inspection of the real filter phase target response data will indicate that the real filter showed a -180 degree shift at 10.35 kHz, and a -360 degree shift at 17.29 kHz. Things get worse from there on out; the real filter exhibits more than 1.5 full phase rotations within its pass-band.

The purpose of this exercise is to be able to synthesize a "fantasy filter" model whose behavior is sufficiently like that of the empirical model to be used as a substitute for it in the ensuing calculations. The object here is *not* to arrive at a filter that is even realizable physically.

It seems that many of the objectionable results of anti-aliasing filter techniques necessary in digital audio technology can be attributed to signal delays that vary with frequency through the pass-band of the filter, because of the progressively greater phase lag with frequency that is a necessary consequence of the extremely sharp filter response slopes[5].

As can be seen in the data given in Table 1A for both the empirical and model filters, the phase-lag through the pass-band continues to lag at an increasing rate with frequency. The consequence of this fact is the relatively

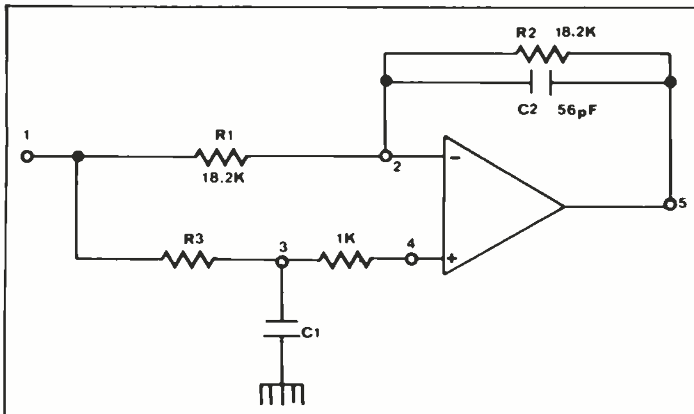


Figure 8: single-pole, all-pass filter configuration.

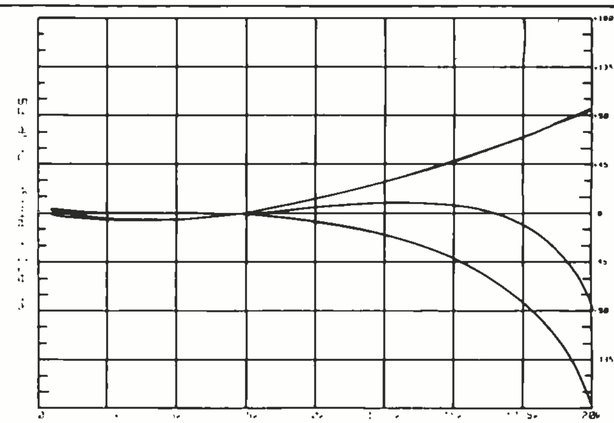


Figure 9: Relative Phase response. Bottom curve: model filter; top: optimized all-pass filter; center: corrected system.

greater time delay of higher frequencies, while lower frequencies are delayed less. One can visualize a group of signals of differing frequency entering the filter at its input simultaneously, and emerging dispersed in time at the output. Which is the cause of the ragged-looking response the filter has for a square-wave input signal, as shown in Figure 2. Figure 3 represents a plot of the normalized 1 kHz square-wave time domain response resulting from the transfer function of the 24th optimization trial filter. Although the detail of Figure 2 shows the general character is there as a demonstration of the power of this method.

The optimization took the computer, an HP 9845B, about eight hours of crunch and grind to accomplish, after several unsuccessful initial trials of circuit models and different sets of pass-band target data. It seems that if the starting approximation of the filter is too far from the final result, and/or if the data points used as targets are too far dispersed in critical parts of the pass-band, the results can be rather deviant from what we would desire as a final result. Certainly, eight hours is a long time to wait for results. For this situation there was no other easy approach to obtain the necessary filter data. In normal circuit synthesis situations, where one proceeds from the other direction and already knows the nature of the circuit topology to fairly high degree—and for which the optimization involves only a few components and a smaller field of target data points — the time required generally is only a few minutes.

Figure 4A shows the magnitude response of the filter circuit used as the basis for the eight-hour optimization. Figure 4B shows the Absolute phase response of the same filter. Note that the first 180-degree lag occurs beyond 11 kHz, where our real filter showed the 180-degree shift at 10.35 kHz, and lagged to 360 degrees at 17.29 kHz. The initial model filter shows a -360 degree occurring at about 18 kHz; not quite close enough for the purposes at hand.

Manual synthesis of a filter having these phase and amplitude characteristics to any accuracy at all would have taken this writer a whole lot longer than the time taken by friend computer. If the truth be known, I would never attempt solution of this kind of problem without very high-powered numerical assistance.

TABLE 1B: MAGNITUDE AND ABSOLUTE PHASE DATA OF THE 24TH ITERATION OPTIMIZED "FANTASY" FILTER AND REAL FILTER TARGET DATA.

Voltage Gain Magnitude:

Frequency	Actual (dB)	Target (dB)	Error (dB)
1.00 kHz	-6.45 dB	-6.80 dB	347.35 mdB
1.60 kHz	-6.42 dB	-6.75 dB	330.73 mdB
2.00 kHz	-6.43 dB	-6.75 dB	321.67 mdB
2.50 kHz	-6.45 dB	-6.70 dB	249.12 mdB
3.15 kHz	-6.48 dB	-6.70 dB	215.93 mdB
4.00 kHz	-6.52 dB	-6.70 dB	184.02 mdB
5.00 kHz	-6.52 dB	-6.65 dB	131.59 mdB
6.30 kHz	-6.46 dB	-6.60 dB	142.06 mdB
7.50 kHz	-6.37 dB	-6.60 dB	232.21 mdB
8.00 kHz	-6.33 dB	-6.60 dB	266.36 mdB
9.00 kHz	-6.29 dB	-6.60 dB	307.55 mdB
10.00 kHz	-6.30 dB	-6.60 dB	300.81 mdB
12.50 kHz	-6.40 dB	-6.60 dB	202.98 mdB
15.00 kHz	-6.35 dB	-6.60 dB	245.09 mdB
17.00 kHz	-6.40 dB	-6.70 dB	301.34 mdB
18.00 kHz	-6.44 dB	-6.60 dB	160.48 mdB
19.00 kHz	-6.58 dB	-6.55 dB	-26.43 mdB
20.00 kHz	-6.54 dB	-6.60 dB	61.34 mdB
21.00 kHz	-14.69 dB	-14.50 dB	-185.08 mdB

Voltage Gain Phase:

Frequency	Actual (deg)	Target (deg)	Error (deg)
1.00 kHz	-14.08 deg	-12.75 deg	-1.33 deg
1.60 kHz	-24.77 deg	-23.60 deg	-1.17 deg
2.00 kHz	-31.61 deg	-30.60 deg	-1.01 deg
2.50 kHz	-40.00 deg	-38.90 deg	-1.10 deg
3.15 kHz	-50.74 deg	-49.80 deg	-935.62 mdeg
4.00 kHz	-64.65 deg	-64.60 deg	-45.29 mdeg
5.00 kHz	-81.06 deg	-80.50 deg	-557.65 mdeg
6.30 kHz	-102.95 deg	-102.90 deg	-47.98 mdeg
7.50 kHz	-124.17 deg	-124.80 deg	631.91 mdeg
8.00 kHz	-133.37 deg	-134.40 deg	1.03 deg
9.00 kHz	-152.40 deg	-153.40 deg	1.00 deg
10.00 kHz	-172.21 deg	-173.20 deg	992.10 mdeg
12.50 kHz	134.87 deg	134.60 deg	267.61 mdeg
15.00 kHz	72.40 deg	71.50 deg	903.22 mdeg
17.00 kHz	9.59 deg	9.50 deg	85.93 mdeg
18.00 kHz	-29.29 deg	-28.40 deg	-892.97 mdeg
19.00 kHz	-76.60 deg	-75.80 deg	-798.96 mdeg
20.00 kHz	-144.02 deg	-145.00 deg	977.90 mdeg
21.00 Khz	115.14 deg	114.80 deg	340.98 mdeg

Group Delay Response

I feel it is important to emphasize that an analysis of the group delay properties of the filter is absolutely necessary to the definition of the problem to be solved in this example. The final resulting fantasy filter performance is compared to the real filter-derived target data as shown in Table 1B. Note that final amplitude response errors are less than ±0.35 dB, and Absolute phase response errors are less than ±1.3 degrees.

In practice, the measurement of group delay is a complicated business. The majority of instrumentation collections generally available will not easily yield such data. The system transfer function as a frequency-domain plot or tabulation of gain and Absolute phase is, in fact, fairly easy to arrive at. Possession of a good phase meter is a prerequisite, but those are rentable if you don't happen to have one on your bench.

By way of yet still further demonstration of the validity and precision of this particular optimization, Figures 5 and 6 show the magnitude and phase response, respectively, over a passband of 1 Hz to 30 kHz. The double traces, in each case, show a 10-trial, ±2% tolerance run, again demonstrating the agreement between the empirically-derived data points, and the 2% tolerance bands of the optimized filter model.

The *real* purpose of all the effort up to this point is to be able to arrive at what we've been looking for all along: the Absolute Group Delay curve of our real filter, shown in Figure 7. The analysis technique involves using a computer-derived linear frequency versus Absolute group delay display for a determination of the region where the group delay is relatively constant with frequency. Having determined the value of the filter group delay in this region to be about 45.4 microseconds, we can then generate a graphic plot of the Relative phase response of the fantasy filter resulting from our many iterations; this Relative phase plot is shown in the lower-most curve of Figure 9. The Relative phase starts out at about zero degrees, and lags at an ever increasing rate as the driving frequency increases. If we can devise some means of providing a system whose Relative phase response is one that leads at an ever-increasing rate with increasing driving frequency, we may be able to cancel out much of the troublesome Relative phase lag of the filter we are bound by the defined limits of this problem.

There is a solution to the constant-time delay problem. The basic element, called a first-order, all-pass filter or, sometimes, a phase-lag network, is shown in Figure 8[6,7,8]. The success of our strategy of generating a network having an ever-increasing leading phase characteristic depends on an obscure property of this all-pass filter. Although the classic all-pass filter does have a lagging Absolute phase response with increasing frequency, it does so in a way that approaches -180 degrees as a limit at some very high frequency. The all-pass network exhibits a constant Absolute group delay characteristic below some frequency, dependent upon one's choice of component values. If this Absolute group delay value is used as the correction value for the Relative phase plot, it will be seen that the Relative phase characteristic of the all-pass filter actually *leads* the input by an increasing amount as the driving frequency rises.

Therefore, it would appear that the installation of a properly chosen all-pass filter in cascade with the output of our fantasy filter should enable us to cancel out the Relative phase lag introduced by our problem filter. The addition of the all-pass network in this way will result in an increase of signal propogation time through the entire system, but the various frequency components of the complex audio signal will have some chance of emerging at the output port at approximately the same time.

With a little application of judicious choice of RC values for our initial lag network, we should be able to provide a complementary phase lead that will act to accomplish the task at hand. The question becomes, then, what are the component values that will result in the optimal cancellation of the anti-aliasing filter's Relative phase lag? The answer to this question and, indeed, many others, can be determined using the COMTRAN Optimization feature.

First, the amplitude and Relative phase data are determined by use of the 45.4 microsecond delay correction constant in the region of constant group delay of the optimized low-pass filter to be compensated. This computation is achieved with AC-CAP single-frequency function, and the data verified using the Relative phase graph plot. The fantasy filter Relative phase data are given in Table 2A. These points are

TABLE 2A: FANTASY FILTER RELATIVE PHASE VERSUS FREQUENCY AND MAGNITUDE DATA TO BE USED AS TARGET DATA FOR THE DELAY CORRECTION FILTER OPTIMIZATION.

Sweep	:Linear
Start Frequency	:0 Hz
End Frequency	:20 kHz
Delay Correction	:45.4 microseconds
Output on	:CRT
Intervals	:8
Voltage Gain is	:Source Gain

Frequency	Magnitude	Relative Phase	Phase Delay	Group Delay
2.50 kHz	-6.45 dB	861.15 mdeg	956.84 ns	800.64 ns
5.00 kHz	-6.52 dB	663.31 mdeg	368.50 ns	548.31 ns
7.50 kHz	-6.37 dB	-1.59 deg	587.63 ns	5.18 us
10.00 kHz	-6.30 dB	-8.77 deg	2.43 us	10.71 us
12.50 kHz	-6.40 dB	-20.83 deg	4.63 us	17.35 us
15.00 kHz	-6.35 dB	-42.43 deg	7.86 us	33.14 us
17.50 kHz	-6.41 dB	-82.95 deg	13.17 us	64.00 us
20.00 kHz	-6.54 dB	-177.13 deg	24.60 us	233.46 us

TABLE 2B: DELAY CORRECTION FILTER RELATIVE PHASE RESPONSE AFTER OPTIMIZATION. NOTE THAT TARGET RELATIVE PHASE DATA IS THE SAME VALUE AS FOR REAL FILTER RELATIVE PHASE RESPONSE OF TABLE 2A, WITH SIGN REVERSED.

Delay Correction : 45.4 microseconds
Phase Weight : 1

The Target Data Is:

Frequency	Relative Phase (deg)
2.50 kHz	-861.15 mdeg
5.00 kHz	-663.31 mdeg
7.50 kHz	1.59 deg
10.00 kHz	8.77 deg
12.50 kHz	20.83 deg
15.00 kHz	42.43 deg
17.50 kHz	82.95 deg

noted for later entry as target data points for optimization of the all-pass correction network. The target data is entered into the Optimization routine for the all-pass filter, with the phase data having the opposite sign of that of the true fantasy filter Relative phase response. This is done because it is the Relative phase response of the fantasy filter that we wish to cancel out; therefore we wish to obtain a network whose Relative phase response is a leading one with frequency at a rate exactly opposite that of the fantasy filter. For the case of the all-pass filter, the gain is substantially unity over a wide passband from DC, so the gain part of the target data can be zero decibels throughout.

All of this computation takes place after the entry of the circuit model of the all-pass filter using values for R3 and C1, as determined in the equation below, based on the relation given in reference #5:

$$C = 1/(2\pi Rf) \times \tan(-tf\pi)$$

Where: t = desired delay time in seconds; -45.4 microseconds in this case.
f = lower band edge of the constant filter group delay region; taken as 2.5 kHz.
R = value of R3, in ohms, taken as 8 kohms.
C = value of C1, in Farads.

The value C, or C1, then works out to be about 2.9 nanoFarads (2.9 nF) as an initial trial value.

Relative versus Absolute Phase Response

Illustration of the essence of what was to be accomplished here can be explained by reference to Figure 9, which shows a plot of the Relative phase characteristics for three cases.

The first case is the bottom curve that intercepts the -180 degree point at about 20 kHz, and is the Relative phase characteristic of the fantasy filter with the 45.4 microsecond Absolute group delay time converted to electrical degrees, and subtracted from each Absolute phase value before plotting as the Relative phase value shown. The top curve is the Relative phase properties of the all-pass delay correction network, which can be derived, again, by subtracting the all-pass Absolute group delay time, converted to electrical degrees at that frequency, from the Absolute phase value of the corresponding point on its Absolute phase curve. The center curve is the Relative phase response of the fantasy filter and the all-pass in cascade, corrected for their combined (45.4 + 33.071) microsecond Absolute group delay.

The words "Relative" and "Absolute" are very important to keep separate. (The term "Absolute phase," used by some to refer to what I insist upon calling "polarity," means to the phase relationship of the output signal of a system referred to the signal at its input as used here.) *Absolute* phase, meaning "absolute time difference" as used in the context of this article, clearly has *nothing* to do with the polarity of the signal, or indeed, of anything else; phase is only another synonym for time[8].

It should be noted that the Absolute phase response of the all-pass is, in fact, lagging, starting at zero degrees at DC and descending to -180 degrees at some very high frequency. The Relative phase plot illustrated only advances in the positive direction because the delay correction time representing the all-pass group delay properties of 33.071 microseconds is being converted to the equivalent phase plot illustrated then advances in the positive direction because the delay correction time representing the all-pass group delay properties of 33.071 microseconds is being converted to the equivalent phase angle at each frequency point, and subtracted from the Absolute phase to yield the points forming the trace shown. This is the end that is to justify these means.

Still referring to Figure 9, it will be noted that the total Relative phase response is fairly flat and close to zero degrees below a frequency of about 17 kHz. It is the closeness of complementary fit of the two individual filter Relative phase responses that make this correction approach work at all.

This example should adequately serve to illustrate that it is not politics that is the true art of compromise. It is *engineering*.

References:

1. *The Book of Apple Software 1983*; et al; ISBN #0736-2692; J. Stanton, R. Wells, S. Rachowansky; The Book Company, 11223 S. Hindry Avenue, Los Angeles, CA 90045.
2. *TRS-80 Applications Software Sourcebook, Vol. IV*; Radioshack Catalog No. 26-2114; pp. 335-6.
3. *Users' Library of Contributed Programs for the HP-41, HP-67, and HP-97, March, 1983; et al:* Hewlett-Packard Co., 1000 N.E. Circle Blvd., Corvallis, OR 97330.
4. *COMTRAN*; D. Jensen; Jensen Transformers, 10735 Burbank Blvd., North Hollywood, CA 91601.
5. *Time Correction of Anti-Aliasing Filters Used in Digital Audio Systems*; J. Meyer; AES Preprint No. 1911 (E-7) October 23-27, 1982.
6. *Handbook of Operational Amplifier Circuit Design*; D.F. Stout, M. Kaufman; McGraw-Hill, 1976; pp. 26-3,4.
7. *Applications of Operational Amplifiers. Third Generation Techniques*; G. Graeme; McGraw-Hill, 1973; pp. 102,3.
8. *Linear Phase Active Analog Filters with Equiripple Passband Responses*; G.D. Tattersall; *IEEE Trans. on Circuits and Systems*, Volume CAS-28, No. 9, September 1981, pp. 925-927.

In the conclusion of this article, to be published in the December issue, Peter Butt will consider COMTRAN's optimization process of circuit delay times, time- and frequency-domain transformations, and real-world circuit response (the latter using a digitized live cymbal crash). As will be seen from the data and graphs to be presented in part two, computerized modelling can provide an extremely valuable analytical tool to the circuit designer — Editor.

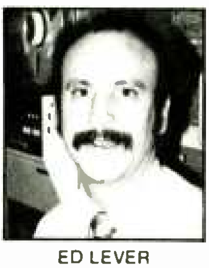
Photography by Kathy Cotter and Debbie Leavitt

PRACTICAL APPLICATIONS OF SMPTE TIMECODE IN AUDIO, VIDEO AND FILM POST-PRODUCTION

by Adrian Zarin and James Riordan

With the development and introduction of digital SMPTE timecode, the Society of Motion Picture and Television Engineers touched off a technological revolution that today still is in its embryonic stage. SMPTE timecode facilitates easy synchronization of audio, video, and film transports. Although intended initially for video editing and post-production, timecode soon found many useful applications in other fields. Indeed, television, film, and audio professionals all developed their own areas of expertise in the use of timecode. In recent years, these different segments of the industry have been brought into closer collaboration, largely due to economic and artistic developments that have taken place within the music business. The result has been some stunning custom applications of timecode. Indeed, it would appear that the prospect of custom interfacing, via SMPTE timecode, of equipment previously thought of as incompatible is just beginning to take off as a practical reality.

"Over the years, we will see a great variety of custom applications of SMPTE timecode," states Ed Lever, president and founder of Los Angeles' Canyon Recorders. Lever has been responsible for some of the more adventurous applications of timecode over the past several years, including the soundtrack for The Band's farewell movie, *The Last Waltz* [described in detail in the August 1978 issue of *R-e/p — Ed.*].



ED LEVER

"There are advantages," Lever continues, "to being able to interlock different forms of sound and picture that only now are being realized. Each new application will produce a new effect that will, in many cases, lead to another application. We're in the early stages, but fortunately the SMPTE equipment and techniques now in use are readily adaptable to future industry needs."

"Obviously there have been new developments," concedes Guy Costa, manager of Motown/Hitsville Studios, Hollywood, considered by many to be one of the industry's best-equipped synchronization facilities. "But the question is: Have there been any new developments in the general use of the code, or has it been more a case of specific developments for specific requirements? I don't really feel there's been any recent novel or unique applications within the use of the code itself at the general level. But there have been some interesting developments for *custom* applications."

GUY COSTA

Timecode Basics

To review some basics, SMPTE timecode is a system for precisely locating and keeping track of discrete points on a length of audio or video tape, and synchronizing each numbered address to the corresponding address on one or more other pieces of tape. It is a digital code that's recorded serially on to the tape, much in the same manner that an audio signal is recorded. Eighty bits of SMPTE timecode information are recorded in sequence on approximately half-an-inch of tape at 15 IPS, corresponding to a time span of about 1/30 of a second. Timecode is generated (encoded) and read (decoded) by a sophisticated range of devices developed by a handfull of manufacturers (see accompanying sidebars to this article).

SMPTE timecode is a Binary Coded Decimal (BCD) system, whereby bits for the numerical values 1, 2, 4, and 8 can represent frame numbers, minutes and seconds of clock time, or a number of other things. A combination of the second and fourth bits, for example, would signify frame #6; 1 and 8 signify frame #9. In addition to these dedicated bits, some "blank" user bits are included at certain points within the serial code; recently these user bits have become a field of experimentation for designers of custom software applications of timecode.

Interface designers have had to deal with the fact that SMPTE timecode exists in several formats, corresponding to the nature and requirements of the different media that use it. The original SMPTE timecode was established for monochrome video, which runs at a speed of 30 frames per second (FPS), and was based on a 60 Hz reference frequency. Each frame of the monochrome video tape has an odd and an even field, each field corresponding to one complete 60 Hz cycle necessary to produce half of a full 525-line video picture.

With the advent of NTSC color television, it was found that color videotape could not operate at the 60-Hz field rate of monochrome video, because of color subcarrier frequency problems resulting in picture distortion. As a result, a slightly lower frequency, 59.94 Hz, was established as the reference standard for NTSC color video recording. In order to reconcile color video's slightly slower running speed with SMPTE timecode's 60-Hz/30-FPS reference frequency (based on the monochrome field rate), it became necessary to devise a new code. For this second timecode standard, the first two frames of each minute of tape, with the exception of every 10th minute, are ignored, or dropped; hence the name for this second SMPTE timecode — Drop-frame. (For the mathematically inclined, 59.94 and 60 Hz timecode differ by about 3½ seconds per hour.)

Often a source of confusion for SMPTE novices, Drop-frame code is more easily understood in the terms offered by Rodney Pearson of Audio Kinetics, manufacturer of Q.Lock synchronizers.

"In order to make your SMPTE timecode frames match the clock on the wall," Pearson explains, "you have to eliminate the first two frames every minute. All you are really doing is changing the *names* of the frames. Basically, the generator that is putting out these numbered frames doesn't "manufacture" the two frames. In other words,

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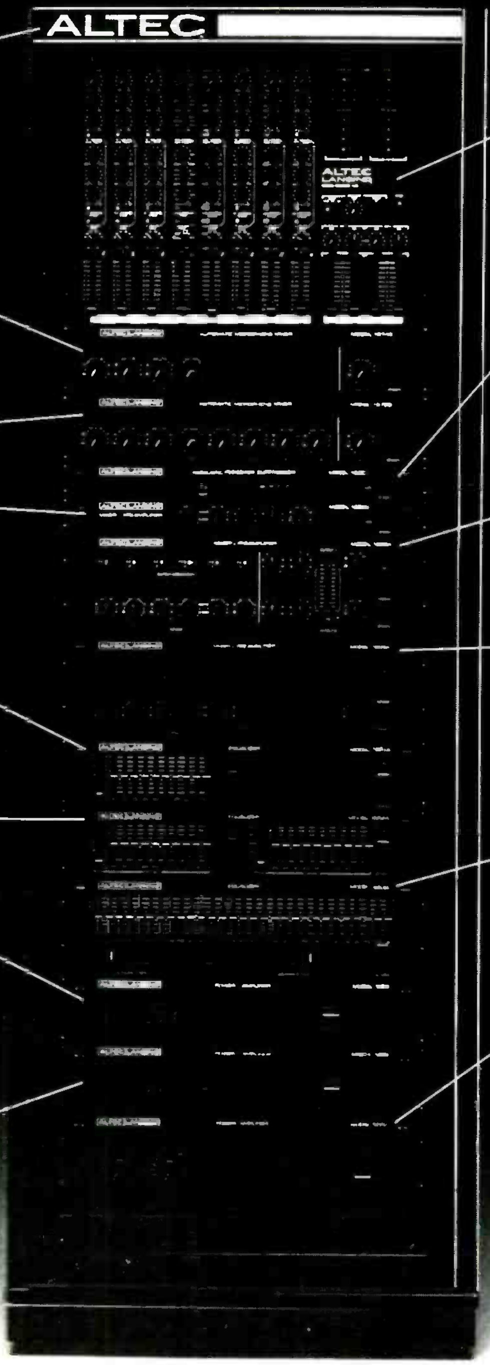

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you're *not* eliminating any audio or video; you're only *changing* the names of the frames."

Apart from the standard 30 FPS and 29.97 FPS Drop-frame code formats, and a few film formats which we will get to later, a further distinction must be made between the *manner* in which SMPTE timecode is printed on different media — i.e. audio versus videotape. As will be readily appreciated, timecode is printed longitudinally on audio tape, usually on track #24 of a multitrack. That is, it is recorded continuously along the length of the tape with no breaks in the code, the 80-bit code printed in this manner is known as Longitudinal Timecode (LTC).

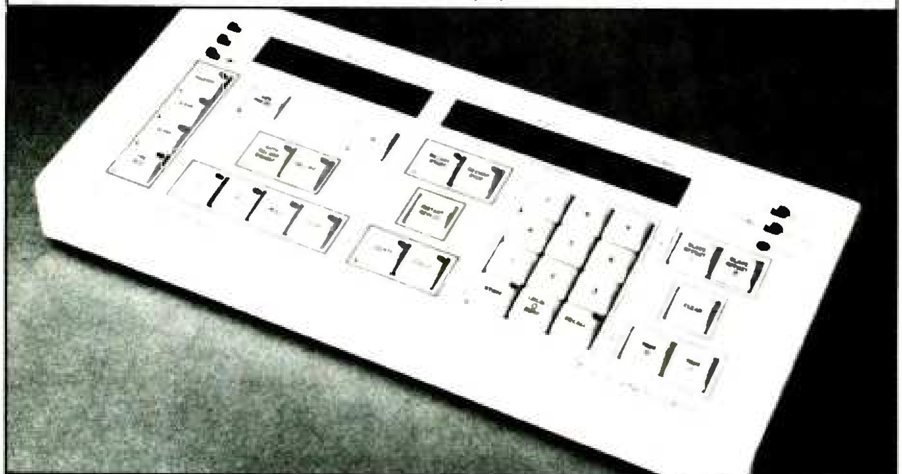
While timecode also can be printed longitudinally on the cue tracks of a videotape, there are occasions when it may not be printed continuously. Since a video picture is made up of two fields per frame, there will be a "blanking interval" —comprised of some 20-odd lines of the video picture — during which the electron beam "writing" the picture across the face of the video monitor/receiver is moved back to the top of the screen to begin the next field. Often this blank interval, which obviously is not used for video information, provides space for a 90-bit timecode called, appropriately enough, Vertical Interval Timecode; many SMPTE synchronizers can be set up to recognize VITC, and handle mixed code formats from audio and video transports.

One of the operational advantages claimed for VITC is that timecode can be read in video still-frame mode. Also, no additional videotape track is needed to contain the code and, since it is recorded along with the video information, it will automatically follow when ever the picture is transferred and/or edited. While early VITC code readers could not handle code at more than a few times play speed, current units are said to be able to follow code at 40 times play speed. And if faster reading speeds are required, a combination code reader — which utilizes LTC for high-speed search, and VITC for play speed lock-up — might be appropriate for certain video applications.

AUDIO KINETICS Q.LOCK 3.10 AND Q.SOFT SOFTWARE OPERATING SYSTEMS

The Audio Kinetics Q.Lock 3.10 synchronizer provides control of up to three audio/video transports via EBU (25 FPS), 30 and 29.97 FPS (drop-frame) SMPTE, and 24 FPS (film) timecode; the system "master" can be redesignated without need for complex replugging. A built-in code generator is capable of 24/25/30 FPS operation, while a Genlock option also is available for code regeneration. Autolocator functions include cycle between nine memory locations, instant replay, locate-to-cue and play, and so on, with or without timecode reference. Machine interfaces can be supplied for a wide range of audio, open-reel and cassette video transports, plus 16/35mm sprocketed film drives.

Built-in RS-232/422 serial ports enable connection to audio console automation systems, and external computer capability. Also, five relay closures can be established by user programmable timecode addresses. To obviate the need for high-speed tape head contact, and 70-kHz bandwidth reproduce amplifiers, tachometer pulses are used to establish transport location in wind mode, and timecode for accurate parking and play synchronization.



Currently available software options for the Q.Lock 3.10 includes Q.Soft SFX for sound effects assembly, which configures the 3.10's control unit to simplify and automate the spotting and assembly of sounds from library tapes and other sound sources (Foley and live-action audio, etc) to a multitrack running in sync with a film or video picture; Q.Soft ADR (automatic dialog replacement) which features 20 cue point memories (or 10 loop cycles), keyboard-modifiable "beep-beep" timing of performers' cues, automatic retention of last loop entered or performed (loop loading automatically sets record start/stop and beep start points), and timecode or feet/frames address entry; Q.Soft VAPP (video audio post-production) for which the video master (usually a U-Matic deck) can be deselected to enable slave #1 (a multitrack containing the music, effects, and dialog to be added to the video) to assume a "sub-master" status to which slave #2 responds (the latter transport serving as the two- or four-track mastering machine); Q.Soft CONFORM, which automatically conforms an audio tape to match an edited videotape; and Q.Soft OPTION 64, which allows the user to instantly select between operating programs, such as ADR or SFX, and to further refine or customize operating procedures to suit a user's specific requirements. In addition, Q.Link Data Interface allows an external computer to access and control all Q.Lock functions via RS-232/422 serial communication links. The external computer can be another 3.10 — enabling five-machine lock-up via a single central control unit —or, for example, a mixing console automation system. Further additions to the Q.Soft family will include Edit Decision List management, and CP/M operating software. □□□

Custom Applications

For technicians needing to work with material in both LTC and VITC formats, devices such as the Adams-Smith Model 2600 Time Code Reader and Translator Interface can provide a convenient solution. But recently, Stevie Wonder's Los Angeles studio, Wonderland, in conjunction with Adams-Smith, came up with an interesting custom application of the Adams-Smith 2600 System. Wonderland wanted to use its new 3M DMS 32-track digital multitrack with its Neve NECAM computer-aided mixing system. The digital multitrack only generated LTC at play speed, however, while the NECAM system required LTC at all speeds for tape cueing and parking. (The reason is obvious, since 3M elected to record the SMPTE code track digitally, which hence cannot be read outside of play speeds.)

"Everything you do on the console is referenced to the timecode stripe on the tape," explains Wonderland's technical director, Lon Neumann. "The NECAM system will send out to [floppy] disk the fact that you moved such and such a fader at such and such an address on the tape, as indicated by the timecode. Well, that's all well and good at play speed on the 3M recorder; it's just a normal situation as it would be with any analog machine.

"But there are several functions that the NECAM performs where it needs to have timecode at fast foreward or rewind. Now, with the digital 3M machine, the phase lock loop or window of the digital system is limited — you can only go so fast or slow, and still be able to read the digital timecode back off tape. Whereas with an analog machine, as long as you have wide-band amplifiers that are able to read the timecode, you can go quite a bit farther in speed.

"Basically, with the digital recorder, you can just read the timecode right around play speed. So rather than give you garbage at fast foreward or rewind, the machine just mutes. The output of your timecode channel is not available for anything, but the NECAM system needs it."

The solution, Wonderland engineers discovered, was to regenerate timecode at the problem speeds using Adams-Smith equipment (normally used for translating VITC), and then feed that regenerated code to the NECAM system. A 2600 Reader Module was connected to the 3M multitrack, which generates tach pulses via a rotating optical wheel attached to an idler on the tape machine. The tach pulses served as the

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Canyon's Edit 1 houses a Grass Valley 1600-3K video switcher, E-Mem effects memory unit, Harris Epic video editing system, and Quantum QM-8P console. SMPTE timecode locations from the Epic computer control all video transport and switcher commands, plus audio levels via VCA channels.

basis for regenerating timecode at fast forward and rewind speeds.

"Through some very sophisticated software," Neumann explains, "credit for which is rightfully due to the software people at Adams-Smith, they generated what results in regenerated timecode that is either incremented or decremented according to the direction-sense signal coming from the 3M machine. So the NECAM thinks it is getting timecode from the tape, but in fact it is regenerated timecode via the 2600 System.

"The code the NECAM gets is correct; its not arbitrary. The Adams-Smith takes the last known good address and, if you're rewinding, for example, it will start counting down from there with the tach pulse. As the tach pulses get faster and faster, the timecode starts counting down faster. Thereby, the NECAM can keep track of the tape at fast shuttle speeds.

As is often the case with custom applications of this nature, the Wonderland/Adams-Smith system may have uses that go beyond the original scope of the modification. "The same type of system we put together here," Neumann suggests, "would be useful for people with conventional analog-type recorders as well, and who find themselves with bad code on tape that they still need to use. This system switches automatically. As soon as it sees that timecode isn't there, for whatever reason — a dropout on the analog tape, for example — it would automatically switch to regenerated timecode and fill in the blanks, so to speak. It will then switch back, smoothly and unnoticeably, as soon as the timecode on tape is good again."

Up until recently, the application of SMPTE timecode in the music recording industry has been confined to a few useful, but relatively modest procedures. Perhaps the most well-known is interlocking two multitrack tape machines via timecode printed on one track of each machine, for control by a single, master machine to yield 46 (or 30) available tape tracks.

SMPTE timecode also has found increasing use as an editing tool for music sessions. A musical passage, such as the bridge or chorus to a song, can be laid off on to a separate piece of tape and then, using the timecode numbers as a precise dubbing guide, "slipped" or repositioned at another part of the song.

"Its being done quite a bit more," says Motown's Guy Costa, "now that you have almost universal use of the Linn Drum Machines and other equipment, where you have sync tracks and can lay the two things together. We have often done that here at Hitsville, but to me it's more of an effect than a creative tool. It's often a matter of getting out of trouble if somebody screws up. Most of our editing, though, is done on a digital editor."

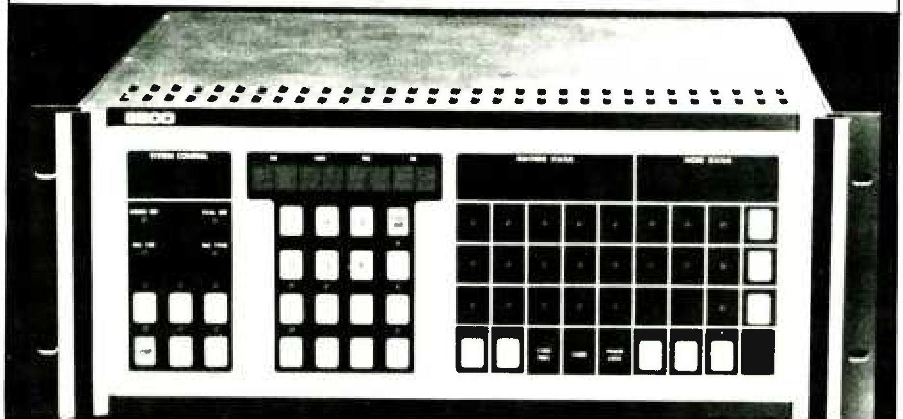
Another musical application of SMPTE timecode is explained by Joel Fein, manager of The Village, a Los Angeles facility that offers full interlock between its four recording studios, theater, and video edit bay, via a choice of Studer, Audio-Kinetics, or BTX equipment. Fein has done such pioneering work as the live soundtrack recording for *The Buddy Holly Story*, and the all-Fairlight CMI soundtracks for Shelly Duval's "Faerie-tale Theatre" cable TV program. He often uses SMPTE timecode as a tool to save wear and tear on recording tape.

JOEL FEIN

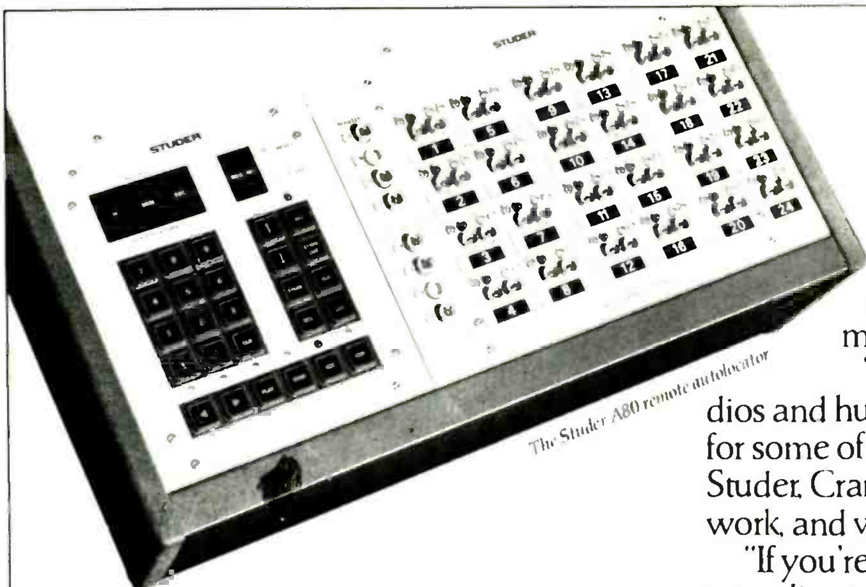
"After you record a basic track," Fein explains, and you're going to start doing something like vocals or synthesizers, you will be making a lot of passes on the tape. So, rather than continually running the master tape, start wearing the oxide off and lose high-end, I'll make a slave copy of the basic tracks — which is

EECO MQS-100A MULTICUE SYNCHRONIZER

The EECO Model MQS-100A will synchronize up to three audio/video transports via SMPTE/EBU timecode (in intermixed drop-frame and non drop-frame formats, if necessary). Two models of the Multicue synchronizer are available: Model MQS-102A contains a main unit electronics, and customer choice of two transport interface assemblies; while the Model MQS-103A comes with a choice of three machine interfaces.



Additional features of the MQS System include the provision of 24-hour offsets between all transports; code-restore circuit for add-on and dubbing applications; timed-event closures for control of external processors and ancillary audio sources; remote operation capability; a "Follow-the-Master" chase capability; built-in scratch pad memories for storing timecode locations entered manually, or on the fly; and an RS-232 UART serial port for external computer control. □□□



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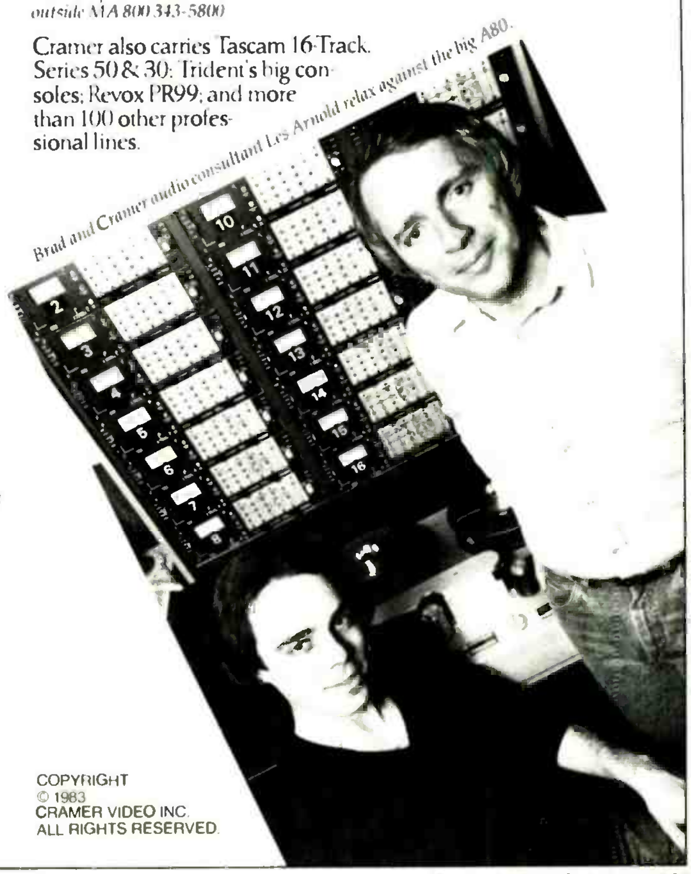


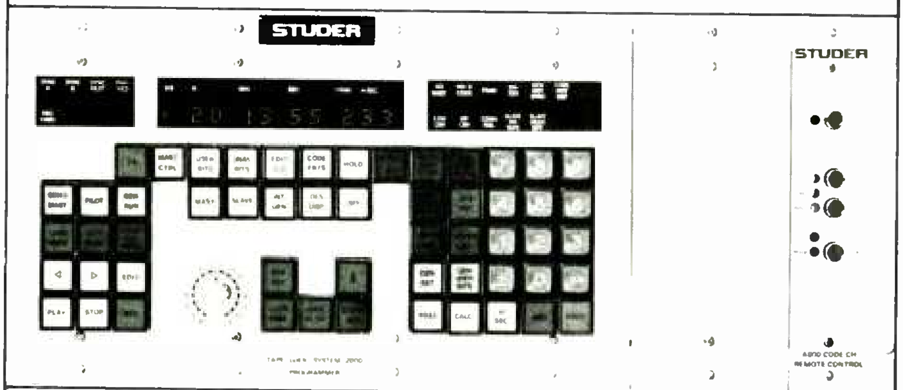
Brad and Cramer audio consultant Les Arnold relax against the big A80.

just everything on the master reduced to two or three tracks on the slave. I'll tie it in with the master using SMPTE timecode [to make the transfer], and then put the master away. We'll go back and do the vocals, synthesizers — everything that's going to take a lot of passes — and put them on the slave.

"Later, we'll tie that back up with the master. You can either transfer back [the slave tracks] to the original tape, or mix with two machines. My preference is to transfer back, and mix on one machine. You can lock the two machines up and have them run together, but then you're still dealing with two discrete 24-track machines. Whereas you may well have room [on the original, master multitrack tape] to combine and bounce down information from the slave. You may have eight tracks of vocals on the slave, for example, which could easily be reduced to two vocal tracks on the master. So you simply mix on to your master, and when you run your mix, you only run one machine. Its simpler that way."

STUDER TLS2000/II SYNCHRONIZATION AND EDITING SYSTEM

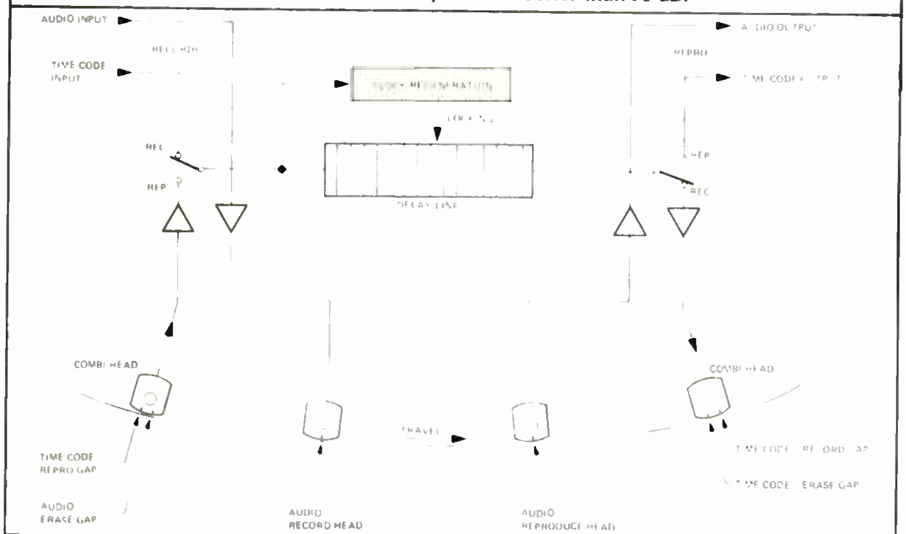
The Studer Tape Lock System 2000 MkII will synchronize two audio/video transports, and provide stop/start control of up to four effects machines against any timecode start location. Synchronization can be achieved against any SMPTE/EBU timecode and user bit information of 24, 25, 29.97 (drop-frame) or 30 FPS formats.



Interface/controller modules are available for both Studer A80 and A800 multitrack transports, as well as for various one-inch VTRs, U-Matic format videocassette, and outboard effects machines. In addition, by cascading two or more TLS controllers, multimachine master/slave combinations can be established.

SMPTE Timecode for Stereo Mastering

The new Studer A810 microprocessor-controlled, quarter-inch stereo mastering machine can be supplied with the capability of printing and reading standard SMPTE timecode on a 0.4mm (0.157-inch) center track between the two audio channels. Specially designed combination heads, placed either side of the audio record and reproduce heads, are employed for recording and reading timecode. A combination head located on the left-hand side of the headblock contains the timecode reproduce gap, and audio erase gap; a second combination head on the right-hand side contains timecode erase and record gaps. Because timecode and audio heads are totally separate, Studer claims a crosstalk rejection of better than 90 dB.



To maintain sync between audio and timecode locations, a special delay line is built into the A810 to compensate for the travel time between the timecode and audio record heads. Similarly, during playback the delay line holds the code signal until it is in exact sync with the audio material. Compensation is automatic at all four operating speeds (3¾ to 30 IPS), even in varispeed mode. Because there is zero time offset between code and audio tracks, offset correction need not be introduced into the synchronization system; also, tapes may be spliced in the normal way without fear of cutting off the corresponding timecode information. □□□

Film and Video Procedures

While for some time SMPTE timecode has had applications like those detailed above in the recording studio, it is the marriage of the latest audio techniques with film and video technology that has brought about some of the most fruitful and resourceful uses of timecode. In reality, such a pooling of knowledge can be attributed to three factors: (1) the overall economic recession in the music industry — and, as a result, the recording studio business — which has led many recording studios into video post-production work; (2) the advent of promotional rock videos; and (3) the growing popularity of rock-oriented films, such as the recent Rolling Stones' concert film, *Let's Spend the Night Together* [see December 1982 and February 1983 issues of *R-e/p — Ed.*] and the highly successful *Flashdance* movie.

"Working with film may indeed be a life saver to many recording studios," according to Canyon's Ed Lever. "There's going to be a lot more film work this coming year as Fox, Paramount, MGM, and many others double and triple production schedules. They're also slashing budgets. No more $20 and $30 million movies. They're trying to keep everything down to between $7 and $10 million, and that means being *real* economical.

"One of the ways film companies can do that is to use more of the audio studios that in the past have only been doing records. Such work can be great for our business at a time when we need it most. The biggest advantage to be gained from an understanding of SMPTE timecode's various applications is for those of us in the music business who want to expand our horizons into other areas.

"There are techniques we have learned in music recording that can be applied very advantageously to film and television, once we understand how their post-production sound is done. We can learn some of their techniques, and embellish upon them by drawing on our experience."

Film work brings with it yet another set of SMPTE numbers. Film runs at 24 FPS, and there is, according to Guy Costa, a 24 FPS SMPTE format timecode. "Twenty-four-frame timecode was developed so that the frame numbers used would be identical to the footage or film frame numbers. If you use standard 30 FPS timecode for film work, there is always a discrepency between the SMPTE code numbers and the film-footage numbers. With 24-frame timecode, however, the code numbers match *exactly* with the film numbers."

"Twenty-four frame is not what I would call an industry standard," Costa

adds. "It has been standardized by use. But I don't think SMPTE acknowledges its existence. Audio Kinetics and Gray Engineering are about the only people [I know of] who make the equipment to run 24 FPS timecode. We are equipped to use 24 FPS code because we have Audio Kinetics equipment."

With this equipment, moreover, Motown/Hitsville also can handle the 25 FPS EBU code, which is based on the European standard PAL and PAL-M 50 Hz frame rate.

Costa prefers not to work with the 24 FPS code, and indeed notes a tendency in the industry away from it, because the slower frame rate makes for less precision in editing. "Muting and things are not as . . . 'tight' is the word, I guess. You've only got 24 frames per second as opposed to 30, and even 30 is marginal when you start doing pickups and cleaning tracks, and things like that. Also, your automation is that much slower on the 24 FPS system."

There are a number of alternatives to using 24 FPS code. One is simply to raise the standard frame rate for film. "I've heard a number of people saying that they're going to go up to 30 frame-per-second film," Costa offers. "In fact, a number of commercials that have come through here have been at a 30 FPS rate. It's just a different linear speed."

Also on the horizon are several techniques for printing timecode directly on to the unused areas of the film itself. Depending on what type of film stock is used, the code could be laid in the space either just outside or inside the sprocket holes. Kodak is developing a system that puts a magnetic coating containing timecode over the film stock. There also is a system, known as Digital Flourescentsound, which involves an invisible flourescent timecode layer superimposed on the film stock, and shows up under ultraviolet light. [See page 126 of the October 1982 issue of *R-e/p* for further details of this digital sound development — *Ed.*]

The most common procedure at this point, however, is to transfer the filmed image to videotape, and interlock using the standard SMPTE code for video. Costa indicates that most of Hitsville's film-scoring projects are done this way, and Joel Fein concurs that the situation at The Village is very much the same. "When we score a film using videotape," Fein says, "the film is transferred to videotape, and a lock is established at 30 FPS. When it is transferred back to film, it is transferred back to the 24 FPS format. So we're only working in one or the other [frame rate] at any given time."

Even so, users wishing to keep track of the film frame rate while editing with the standard video timecode rate can do so by employing the user bits left open in SMPTE code.

"In order to do rough-cut editing," Costa offers, "a lot of people are putting information detailing feet and frames into the user bits. They just view the user bits either on an insert mode superimposed on to the picture, or they actually put them into the picture. This allows the feet and frame 'technique,' if you will, to be used in rough cut editing, scoring, and overdubbing."

Customized SMPTE Interfaces

The transferring of film to videotape allowed for some fancy SMPTE footwork on the part of engineer Lee DeCarlo during post production of the *Flashdance* film soundtrack. DeCarlo used SMPTE timecode to interlock two 24-track machines and a ¾-inch U-Matic videocassette recorder, with the Total Recall automation system fitted to one of the Los Angeles Record Plant's Solid State Logic consoles. Interlock was achieved by interfacing three Apple II computers, each one driving one of the audio or video transports, with the SSL automation system.

LEE DECARLO

"We could just press one button," DeCarlo states, "and say, 'go to the first verse,' then all the machines — includ-

PRACTICAL HINTS AND TIPS FOR USING SMPTE TIMECODE

Listed below are some of the more important aspects to bear in mind when working with SMPTE timecode for audio, video, and film production. It contains hints and tips garnered from the various individuals whose comments are included in the accompanying article.

Printing Time Code

The most important parameter to watch for when recording timecode on audio or videotape is to print it at a high enough level to survive dubbing, but yet not so hot that it bleeds on to adjacent tracks. Motown's Guy Costa recommends a level of -10 dB for audio tape, while Joel Fein, of Village Recorders, says that he prints timecode at about -6 dB. The conventional location for timecode is on the last track of your multitrack machine — track #24 or track #16, depending on the format in which you're working. As for printing timecode on videotape, Costa says, "We try to go +3 if possible — almost close to saturation. This is mainly because the [video] editing systems need to have a +3 code level in order to work; it has nothing to do with our equipment. But when we either get material in here at Hitsville, or ship tapes out that have video laid over on it, we have to have as hot a code as we can get."

Printing a Reference Tone

It is practical and customary to print the appropriate field-rate frequency on the track immediately adjacent to the timecode track on audio tape.

"Laying on a 59.94 Hz reference is an *essential* step for using timecode in television applications," according to Costa. "Most recording studios are not equipped with what video people term "house sync," which generates a 59.94 Hz synchronization signal." One solution is to use a field-rate converter which allows you to put a video signal in, and get 59.94 Hz out, Costa explains. "Other products take in a timecode signal and reshape it, giving you back timecode and a 59.94 resolve tone."

"In video work," adds Canyon Recorders, Ed Lever, "SMPTE timecode must be printed so that for every so many control-track pulses, there is one and only one *unique* timecode number or frame. It is very important, therefore, to reference the timecode to a source of synchronization, and to record both the code *and* the reference tone at the same time to ensure reliable, repeatable lockup."

"Our standard procedure calls for 30 seconds of reference pre- and post-roll," says Costa. "This way you can reconstitute the timecode later on if you have to. Also, we always attempt to start at a one-hour rate for the downbeat, to make things compatible with systems like the NECAM [console automation system] that don't go from zero. There is a tendency when we're working on an album to start each tune at a different hour rate, so we can distinguish among the tunes."

Adjacent Tracks

The best procedure, according to Costa, is to always use the track adjacent to the timecode for your video sync reference tone. "Nobody likes to use track #23 anyway, because of problems with the timecode and program material leaking into one another. Laying your 59.94 Hz tone on to the adjacent track will keep everybody out of trouble."

Should you have to lay program material right next to the timecode for some reason, it *can* be done if you exercise caution. Both Costa and Fein acknowledge having run signal against timecode with no problems.

Copying Timecode

"*Always* reshape timecode when you go to make a copy or a slave," Costa advises. "Always go for a reshaper and, as much as possible, use a slew-limited code to minimize crosstalk problems. A common mistake people make is to try and record SMPTE timecode on a master while also making a slave tape, so that both timecodes are first generation. This will not work. *Always* make your master first." □□□

ing the video deck — would go to that SMPTE location, stop, and park. The automation system would then say, 'Ah, we're at the first verse,' and every knob on the console [under control of the SSL Total Recall] could be reset in position for the first verse.

"This was using the Apple computers and the SSL console, which is a disk-based, real-time system. It was unique because nobody had ever been able to interface all these different kinds of machines before. We worked at it a couple of days trying different interfaces, and changing the software around just to get the machines to talk to one another. The one thing that computers *really* like to do is talk to one another. They've got good internal communication because they're all seeing the same signal — the SMPTE timecode. This makes it a lot easier when you want to break it down."

The transfer of film to video, and the use of a custom interface, yield many advantages, according to DeCarlo, for both tracking and mixing. For one thing, such techniques replace the clumsy mechanics of film projection for scoring with the ease of video monitoring. A video monitor was placed right above the console during mixdown of *Flashdance*, allowing detailed panning, volume shifts, and so on to be closely linked to picture. For tracking, DeCarlo says, "you can have a whole bunch of [video] monitors lying around. You can sit one in front of the drummer, one in front of your string section . . . and you don't have to all be facing in the same direction."

Perhaps more importantly, DeCarlo's system drastically cuts down on the rewinding time of 35mm mag dubbers and film projectors. "Hours of studio time are wasted on rewinding film," he says. "The time for one rewind is something like 15 minutes, by the time they get everything cued up and everybody is ready to go. In the time they take to rewind once, we can do three or four different takes just by using a 24-track machine, a television monitor, and a computer."

And such a system, DeCarlo feels, also can be applied outside music recording in such areas as dialog looping where, he estimates, "only 10% of the time is actual performing time — the actor reading his lines. The other 90% is [spent] rewinding, and getting ready."

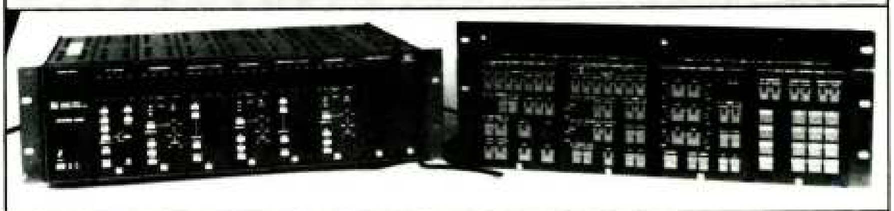
Apart from the convenience, DeCarlo's system allowed him at one point while scoring *Flashdance* to salvage a dance sequence that accidently had been shot to completely the wrong tempo. DeCarlo had sent director Adrian Lyne, who was shooting in New York, some rough audio cassettes of music for demo purposes only. Due to a misunderstanding within Lyne's staff, the audio was transferred to a Nagra portable tape machine using a cassette deck that ran at an inconsistent speed, and film was then shot to replay from the Nagra. As a result, the videotape of the dance sequence that was sent back to DeCarlo for final music dubbing ended about *15 seconds* before the actual 24-track score did!

"You couldn't compensate with the varispeed control on the tape machine," DeCarlo recalls, "because the speed variation on the cassette wasn't constant; it speeded up and slowed down randomly. The problem was to get that random [aspect] to occur with the 24-track machines at the same time that it occurred on the film."

The solution worked out by DeCarlo and soundtrack producer Phil Ramone was quite ingenious. As DeCarlo explains, "I got one of those Clap Trap machines that are on all the new disco records, and recorded it on to one of the three [audio] tracks of the video machine. Every time the snare drum hit on the version of the song that was on the vidoetape, I hit the Clap Trap manually along with it. I recorded the same

ADAMS-SMITH SYSTEM 2600 MODULAR SYNCHRONIZATION SYSTEM

Currently comprising a family of 18 reader/translator/control/interface modules, the Adams-Smith System 2600 can be configured to handle individual or mixed SMPTE (LTC) and VITC timecode and user-bit formats, as well as connection to external computers via serial (RS-232/422) and parallel interface ports. Unlike comparable lock-up systems that can accommodate both LTC and VITC code for video synchronization, but which utilize LTC for actual editing purposes, the company claims that System 2600 enables VITC code to be used for both edit decision *and* execution, including shuttling, cueing, parking, syncing, and locking.



VITC code tracks can be recovered at any tape speed — from still-frame to 45-times play speed — in bidirectional mode; the recovered VITC also can be translated into LTC code and fed to existing electronic video editors. Recovery of conventional LTC tracks is in the range from 1/20- to 100-times play speed, in either transport wind direction.

System 2600 modules measure one or two inches in width by 5¼ inches high, and are 15 inches deep including rear-mounting connectors. Virtually any combination of modules can be arranged to suit the audio/video synchronization task to be undertaken.

thing on to an open channel of the 24-track, again manually reproducing the incorrectly timed drum beats from the videotape. By hitting the Clap Trap manually, and not using the automatic timing on the machine, I got the same timing errors on *both* pieces of tape.

"I then introduced two more computers to the interface, feeding the Clap track beats from the video machine into one computer, and those from the 24-track into the second. The computers measured the time discrepencies between each beat. If beat #22 came five milliseconds later than beat #21, the computer would register that fact, and compute the number of frames by which the beat was off, using the 30 FPS standard. This information was fed into the Audio Kinetics Q.Lock, which in turn governed the machines."

DeCarlo also found it necessary to

Motown/Hitsville's machine room.

modify the Audio Kinetics synchronizer. "On the new Audio Kinetics [Q.Lock 3.10]," he states, there are three channels — a master, and two slaves. SMPTE on the first machine tells the other two where it's at, and the other two chase. I didn't want that though, because there would always be a lag between the master and the slaves [since the latter purposely would be running slow, because of timing differences]. So, in essence, what I did was make it so that there was *no* master, and all three machines became slaves. This way, all three would move at the same time, as opposed to having two chase the other one.

"Guided by the Audio Kinetics, which was getting SMPTE timecode information on all the speed discrepencies on the videotape copy of the film, all the 24-track machines moved with the videotape all the way down the line. *That's* how the real music got synced up to picture."

Timecode Confusion

DeCarlo is an outspoken critic of some of the ways in which SMPTE timecode has been administered up to now. "SMPTE is one of my pet peeves," he says. "There's about 8 million different kinds of SMPTE out there in this world. I've *no* idea why. There is a special Czechoslovakian SMPTE timecode! Now, where did they come up with that? A lot of the people who are setting the standards for things like SMPTE timecode are not the people sitting in the rooms using the equipment. So a lot of unrealistic things go on."

"But SMPTE is our friend," he adds. "It can be used to link everything together with everything else, and make sure it all works properly — as long as you do it with some sort of discretion, and realize that you've got to have some form of standards you can work with."

In DeCarlo's opinion, the audio industry has a lot to contribute to film production techniques. He cites the time his computer system saves on rewinds as an example. "Audio technology is miles ahead of film technology," he concedes. "For the most part, the film industry is using the same equipment and techniques they've been using for 20 or 30 years, and hasn't really moved with the times. But it is still the movie and video people who are dictating what SMPTE

BTX SHADOW II, CYPHER, AND SOFTOUCH SYNCHRONIZATION SYSTEMS

The BTX Shadow II system, comprising the Model 4730 command console and Model 4700 rack-mount electronics unit, will control two audio or video transports, and offers all the features of the Shadow synchronizer, plus, according to the manufacturer, improved code reading and easy front-panel access to all calibration functions.

The Model 6000 Cypher modular timecode system is designed specifically to be interfaced with products supporting an RS-232 serial interface bus, and can be configured to generate, read, display, character generate, and jam-sync SMPTE code and user data; vertical interval timecode (VITC) and "standard" LTC formats also can be accommodated. A companion unit, designated Model 6200, allows the user to generate, read, and jam sync VITC; the unit's VITC reader offers automatic VITC to SMPTE LTC code switching and conversion for editing purposes.

The company's recently introduced Softouch™ System comprises three distributed intelligence modules: the Softouch Controller/Editor, Shadow II synchronizer, and the Cypher timecode system, all interfaced via RS-232 serial busses. The system will control up to four audio/video transports (while supporting additional transports in chase-lock mode), plus a maximum of 16 other devices requiring triggering via timecode-controlled contact closures. Generating or reading SMPTE, EBU (25 FPS), VITC, or 24 FPS timecode, regenerating code or jam-syncing code can be accomplished at the touch of a command key.

Being software-based, Softouch allows frequently performed multistep routines to be committed to a Softkey memory, and executed repeatedly at the touch of a single key. (The user can define, edit, and store as many as 16 Softkeys at a time during a complex post-production sessions.) Sufficient memory is available to contain all pre- and post-roll, plus "beep-tone" trim, mark in/out, and record in/out data for up to 100 loops.

In addition, the system allows separate record assignments for each transport, as well as master record enable; full "wild-machine" control; the ability to autolocate any transport independently of any other activity; and control of the record window during looping functions. □

Canyon's ADR Room centers around a Quantum QM-12D, Ampex MM-1200 and ATR-104, Q.Lock 3.10, and Sony BVU-800.

timecode and things like that are."

Because of SMPTE timecode, it has now become feasible for the musical portion of rock movies and videos to be recorded and mixed in the way this music was intended to be — in an audio recording studio or remote truck — rather than on alien equipment in a facility geared for film or video work. Such was the case with the recent Rolling Stones' concert film, *Let's Spend the Night Together*, directed by Hal Ashby.

The Stones' movie was shot on film, an edit-decision list created via videotape copies, and the final edited film transferred to video, along with a copy of the edit-decision list, to enable the multitrack audio to be conformed to picture.

According to Rodney Pearson of Audio Kinetics, the company which provided custom software for the project, "The soundtrack was actually mixed at the Power Station in New York. They were able to mix the sound to the Stones' live concerts in a recording studio using an engineer, Bob Clearmountain, with whom the band was familiar. It was dubbed at a film studio, but they were able to do the actual mixing under the very circumstances that had prevailed when they did the accompanying live album.

"In the past, the group would have probably had to compromise on the music for the film. This system gave them a great deal more flexibility, and is the kind of thing that will happen as more people become familar with SMPTE applications."

Production of the soundtrack for the Stones movie also called for some custom software applications using SMPTE timecode. According to Pearson, "the production crew [headed by director Hal Ashby] wanted to be able to rewind the film at play speed, and have an audio machine [Ampex ATR-124] run with it in sync. Instead of rewinding at two or three times play speed, which is normally dead time, they wanted to run at play speed and check levels and so forth during rewind. Admittedly, they had a longer rewind time, but they were able to make more use of it.

"Audio Kinetics was able to develop custom software specifically for that application. Custom software is definitely the wave of the future. There is now a wider variety of machines that can be synchronized, and the applications involve multiple systems being used in a variety of ways. That means more custom work, rather than any form of standardization."

A comparison of the different approaches taken to the problem of dead rewind time on the *Flashdance* project and *Let's Spend the Night Together* illustrates the truth of Pearsons' comments. While DeCarlo and the *Flashdance* crew sought to drastically reduce rewind time via the use of videotape and a custom computer interface, the crew working on *Let's Spend the Night Together* used custom software to actually prolong film rewind time, while putting it to good use. As is shown by the many custom applications to date, SMPTE timecode is well on its way to becoming a personal production tool limited only by the user's creativity.

■■■

AUTOMATED STUDIO TECHNOLOGIES SMPTE-BASED EDITING SYSTEM

The AST Editing System, currently being distributed by Harvey Professional Audio/Video, is a SMPTE timecode-based controller for use in either audio or video post-production. Standard features include control of up to 12 tape transports, 999 event list management, and a Library function to store timecode locations of sound effects, theme music, or a shot list. Options include production switcher interfaces and control over as many as 10 outboard audio processors.

Direct serial or parallel control of tape transports is supported. Where no such address ports are available, the controller may be interfaced to the BTX Shadow synchronizer. In this case, varispeed, shuttle, and track assignment switching is provided on machine specific "personality cards."

The controller is described as being especially easy to use and understand. List management, for example, is accomplished by moving an "entry window" through the edit list. The position of this "entry window" on the screen determines whether an in- or out-point is being designated. Since record and play machines are automatically kept in the appropriate column, a single Enter key replaces the usual Record In, Record Out, Mark In, Mark Out entries.

Two-machine record capability permits A/B roll video editing while recording basic tracks on a synchronized multitrack. Video editing facilities now have the option of expanding in-house audio services, or providing clients with first-generation masters for sweetening elsewhere.

The Library feature provides a means to catalog and store SMPTE timecode locations of theme music and sound effects collections. Once a particular selection has been found, its timecode location can be entered into the Edit List by a single keystroke. At the same time, the complete description and reel number are entered into a companion Text file for reference. In this way an engineer can view a video tape or film and accurately "spot" the music and effects to be recorded later. Library selections can be found directly by name, or by searching in an alphabetic directory; in addition, families of related sounds may be organized into Files. □□□

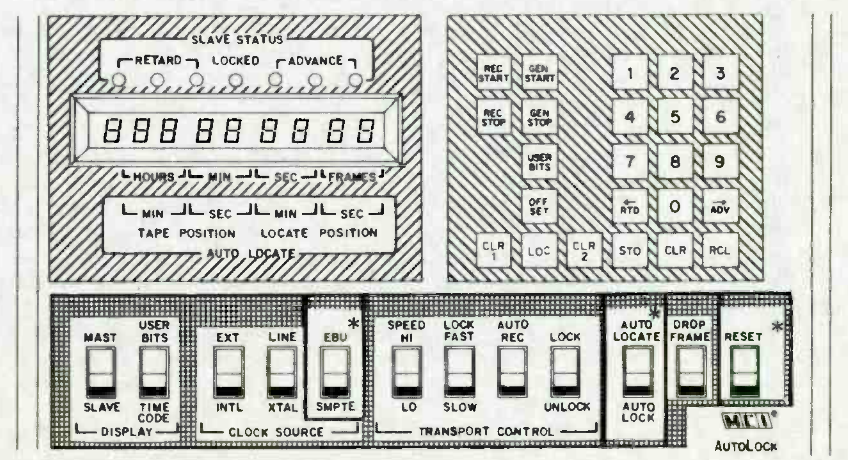
MCI JH-45 AUTOLOCK SYNCHRONIZER

A single-unit SMPTE/EBU generator/reader/synchronizer and AutoLocator, the JH-45 slaves any MCI tape transport to audio/video/film master machines providing a SMPTE/EBU or video "house sync" source. The unit generates SMPTE, EBU, video drop-frame (59.94 Hz) codes, and codes synced to 50/60 Hz external power frequency, composite video signal with selections of odd/even field, and vertical serration user programmable. Also, user bits can be changed as code is being generated, without affecting code continuity, and frame mode slew rate is said to be limited only by fast forward/rewind speed.

By reading either tach pulses, or high-speed timecode for chase function, no wide-band replay amps are required. A code display indicates absolute difference between master and slave, while "park-slave" capability allows the machine to stop within a frame of timecode display. In addition, punch in/out record can be set up at a selected programmable sequence.

In AutoLocator mode the unit provides automatic read/write of tape position counter and 10 memory locations recorded on an unused portion of the tape (approximately 30 seconds) for use in later sessions, without having to enter it manually. Also, a shuttle function is provided between two points.

Compatible and simple interfaces are available for all MCI transports; interfaces for other audio/video equipment can be supplied on request. □□□



New Products

FENDER UNVEILS FULL LINE OF MIXERS, POWER AMPLIFIERS, & MICROPHONES

The first 16 products from Fender's newly created Pro Sound Products division comprise three mixing consoles, five powered mixing consoles, two stereo power amplifiers, and three new series of microphones. "Our objective with this line was not to create another 'me too' series of PA gear," stressed division marketing director Steve Woolley. "We drew upon Fender's strength and experience in equipment for live performance — and approached it from the standpoint that whoever's operating the sound system is giving a performance, too."

According to Woolley, the company used outside consultants, as well as its own engineering talent, to identify key areas for improvement over currently available gear. Examples cited included professional 48-volt phantom powering in all mixers, to allow the use of studio-quality condenser microphones; the use of balanced differential input circuitry to eliminate the distortion associated with typical mike input transformers; and the provision of signal present and peak LED indicators on every input channel to quickly pinpoint problems.

The company's two new power amplifiers — as well as the power stages in powered mixers — are designed to operate with load impedances as low as two ohms. "Typically in the real world, a band will just keep plugging in speaker cabinets until they run out of cords. You have to design amplifiers to handle it," Woolley observes. Particular care was taken in anticipating and eliminating possible problems in interfacing Fender mixers and amplifiers with other electronics, including careful design of each component's internal grounding system, and the inclusion of signal buffers at inputs and outputs.

The new Pro Sound line breaks down as follows:

Powered Mixers

A total of five models with built-in power amplifiers is headed up by Model 3106 six-channel mono unit with 200W output, separate monitor and effects

busses, and a patchable nine-band graphic equalizer. (Suggested MRP is $895.) The remaining four models are stereo units with two, 200W power sections (patchable for stereo left/right, house/monitor, etc.). The 3206 has six inputs, two independent monitor mixing busses, and two patchable graphic equalizers; models 3208, 3212, and 3216 incorporate four graphics and 8, 12, and 16 inputs respectively. Prices range from $1,195 to $2,095.

Stereo Mixing Consoles

Designed for permanent installations or touring setups using separate power amplifiers, the Models 4208, 4212, and 4216 are 8-, 12-, and 16-channel stereo mixers with such features as a signal insertion patch point on each input, four auxiliary inputs (two with panning),

dual monitor mixing busses, a built-in headphone amp, and +24 dBm transformer-isolated line outputs. The top-of-the-line Model 4216 adds a cue/solo capability and switchable high-pass filters on each channel. Suggested retail prices are $995, $1,195, and $1,895, respectively.

Power Amplifiers

Two dual-channel units, designated Model 2244 (440W per channel into 4 ohms) and the Model 2224 (240W per channel), feature variable-speed forced cooling, and high-current output stages

that are said to eliminate the need for triggering of VI limiting in any conceivable application. Electronically balanced transformerless bridging inputs are standard, while a mode switch selects stereo, mono (which allows driving both channels from a single input), or bridging mono, which allows the Model 2244 to develop 84 volts RMS (880 watts) into 8 ohms. A 12 dB per octave high-pass filter may be switched in at 20 or 40 Hz. Suggested retails are $1,150 for the Model 2244, and $795 for the 2224.

Dynamic and Condenser Microphones

The D-Series of three dynamic cardioid mikes feature characteristics tailored to enhance vocals, with a slight presence lift and a predictable bass-proximity effect. Prices range from $70 for the Model D-1, to $149 for the D-3, which is specially designed for applications such as female vocals.

The P-Series of "permanently charged condensers" are said to deliver the wide, flat response and the accurate, neutral sound of studio-quality condenser microphones, plus the ruggedness of a dynamic. The Model P-2 retails for $99, while the P-1, priced at $220, features switch-selectable response tailoring, undistorted 150+ dB capability, and internal battery or phantom powering.

Designed to solve difficult miking problems, such as acoustic instruments and drums, the Model M-1 miniature mike consists of a capsule only slightly larger than a pencil tip. The unit's preamp pack features a tunable notch filter (ideal for suppressing the runaway resonant "boom" on amplified acoustic guitars), a rolloff and on/off switches. Capable of being powered by its own internal batteries, or external phantom power, the M-1's 150+ dB SPL capability enables it to be used inches from drumheads (even kick drum) and cymbals for excellent separation without distortion. The Model M-1 is priced at $175.

FENDER MUSICAL INSTRUMENTS
1300 E. VALENCIA DRIVE
FULLERTON, CA 92634
(714) 879-8080

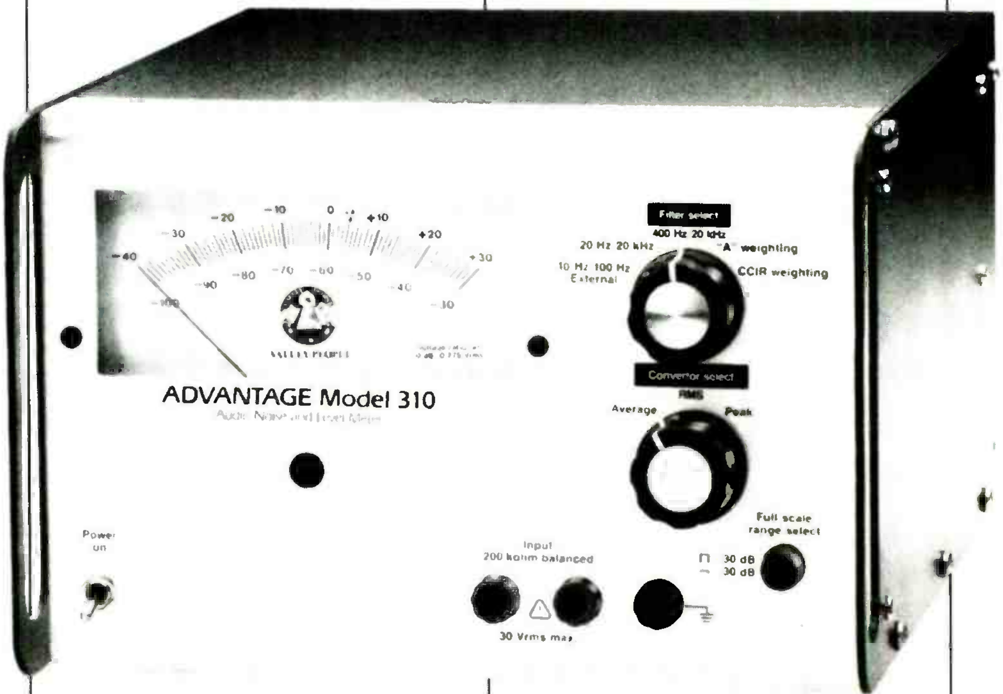
For additional information circle #126

STUDER DEBUTS MODEL 2706 MONITOR SYSTEM

Designed to serve as the primary audio monitoring system in small to mid-sized control rooms, the new Studer 2706 monitor is said to be particularly suited for use in radio studios and television post-production suites, and also may be utilized as a close reference monitor in larger mixing rooms.

A three-way system, the 2706 incorporates a 12.5-inch woofer, a 2-inch dome midrange, and a 1-inch dome tweeter in a bass reflex enclosure. Crossover fre-

In test and troubleshooting, the solution rests with the equipment!



There's no substitute for good logic and sensible measurement techniques when specifying and measuring audio noise and levels. But, without equipment incorporating the specific features required for analyzing noise performance in modern audio equipment, even the most accomplished engineer faces an insurmountable task.

The Advantage 310 Audio Noise and Level Meter was created to provide recording studios and broadcast facilities with a low cost, high quality measurement device capable of delivering greater accuracy than that achievable with more expensive general purpose instruments.

Consider the advantages offered by the Model 310:

Isolated, balanced, Trans-Amp™ differential inputs–To eliminate unwanted noise, RF, and hum pickup while allowing measurement of extremely low level signals.
10 Hz to 100 Hz "wide band" filter–Allows insertion of an external weighting network as required by user's measurement application.
20 Hz to 20 kHz multiple pole filter–To achieve accurate measurement of noise in an actual 19,980 cycle bandwidth through incorporation of 18 dB/octave filters with appropriate −3 dB points.
400 Hz to 20 kHz multiple pole filter–To remove the lower four octaves of the audio spectrum in order to eliminate measurement errors caused by ground loops and other low frequency noise sources.
"A" weighting filter–Correlates noise measurement to loudness as perceived by the human ear.
CCIR weighting filter–Allows noise measurement to be equated to current European standards.
Average dectector response–Indicates reading in commonly accepted volumetric units (vu).

With the Model 310 Audio Noise and Level Meter, the Advantage is in your corner! It has the features you need, at a price you can afford, so you get the sound you desire.

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RMS detector response–To accurately measure the effective energy of steady-state waveforms of varying complexity.
Peak detector response–Allows accurate measurement of the magnitude of waveform peaks.
Large, east-to-read, dual scale analog meter–Accurately displays a wide 70 dB range in 1 dB increments, allowing the user to easily interpret the nature of the noise being measured without changing the range.
Full scale range select–Two ranges allow measurements from −100 dB to +30 dB, thereby encompassing the entire spectrum of aniticipated noise and signal levels common in audio equipment.
Detector output–Provides for indication of input signal level by an oscilloscope or other outboard recording devices.
Preamplifier output/return–Allows connection of monitor amplifier and speakers without interfering with the measurement process.

For additional information circle #127

quencies are 720 Hz and 2.5 kHz. Anechoic chamber frequency response (90 dB SPL at 1 kHz, sinewave sweep) is quoted as 42 Hz to 20 kHz, ±3 dB.

The Model 2706's woofer is designed so that the magnetic field remains constant over the entire excursion range of the voice coil. This is said to ensure very high linearity to just below the clip point, thus reducing distortion. Midrange and high frequencies are handled by newly developed, high-power dome transducers with carefully defined polar patterns. Filter slopes of the crossover network were selected to precisely match the corresponding mechanical properties of the three transducers.

Nominal impedance of the 2706 is 4

ohms, maximum output level 104 dB SPL, and dimensions are 24 by 15 by 13½ inches (H × W × D). Price for the Studer 2706 is $690 each.

STUDER REVOX AMERICA, INC.
1425 ELM HILL PIKE
NASHVILLE, TN 37210
(615) 254-5651

For additional information circle #128

PULSAR 80 AND 40 SERIES MIXING CONSOLES

Total modularity of the new mixer series provides the user option of adding signal processing modules, such as the company's four-channel Comp-Limiter, dual 10-band EQ, reverb and dual stereo mixdown module. Full patching is accomplished by access in and out jacks on all modules.

Both the 80 and 40 Series boards have a flexible matrix mixing capability. A matrix mix allows eight independent mixes of eight groupings to be set up simultaneously and independently of one another. These eight vertical (matrix outs) and eight horizontal (group receives) comprise a total of 64 controls. Each of the volume controls

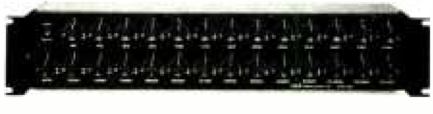
can represent inputs 1 thru 32, providing almost limitless flexibility. For example, if a band can't afford a main and monitor mixer, the main board can double as both: matrix Mix #1 could be the house mix, Mix #2 the side-fill mix, Mix #3 monitor #1, Mix #4 monitor #2, and so on.

PULSAR LABS, INC.
3200 GILCHRIST ROAD
MOGADORE, OH 44260
(216) 784-8022

For additional information circle #129

WHITE ANNOUNCES MODEL 4520 THIRD OCTAVE PASSIVE EQUALIZER

The new equalizer features 27 single-tuned, L-C filters on ISO third-octave frequency centers from 40 Hz to 16 kHz. The filters are individually tuned to a tolerance of ±3% of center frequency, and continuously adjustable to a maximum insertion of 10 dB. All controls are MIL-spec, conductive plastic, rotary potentiometers.



The Model 4520 features two outputs and an accessory octal socket into which optional, low-level crossover networks may be installed for bi-amp operation. The company offers a full line of audio filters and crossover networks, including crossovers optimized for constant directivity horns.

The unit, which weighs only six

pounds and requires only 3½ inches of rack space, has a suggested retail price of $725.

WHITE INSTRUMENTS, INC.
P.O. BOX 698
AUSTIN, TX 78767
(512) 892-0752

For additional information circle #131

SESCOM MODEL ADA-1 FOUR-CHANNEL DISTRIBUTION AMPLIFIER

Packaged in a 1¾-inch rack mount enclosure, the ADA-1 is designed as a four-channel, line-level distribution amplifier, or as four individual line amplifiers. The common input feed is balanced-bridging. A set of unbalanced inputs also are provided by way of four ¼-inch phone jacks that interrupt the corresponding line amplifier and all higher number amplifiers from the common feed.

Specifications include maximum input level of +18 dBv; gain of zero to +30 dB; noise 101 dB below rated output; and distortion less than 0.2% at 20 Hz maximum rated output.

User price of the ADA-1 is $195.

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

For additional information circle #132

SHURE EXPANDS PE SERIES WITH PE86 AND PE66

According to Shure design engineers, the new models offer a level of performance comparable to that of the company's SM microphones. "Every effort has been made to give the PE86 and PE66 the same distinctive, punchy sound that has been the hallmark of Shure's famous vocal microphones," said Travis Ludwig, Shure's technical coordinator. "The new models have the same ruggedness, reliability, and crisp, clean sound."

Both microphones are cardioid patterned, dual-low impedance models with shock-mounted cartridges for quiet operation. The PE86 has a frequency response of 50 Hz to 15 kHz; the PE66's is 40 Hz to 15 kHz. Both models feature a fixed bass rolloff, and an upper mid-range presence peak. The PE86 also features a built-in spherical windscreen to minimize wind and breath noise.

User net prices are $125.00 for the PE86L-LC, and $109.25 for the PE66L-LC.

SHURE BROTHERS, INC.
222 HARTREY AVENUE
EVANSTON, IL 60204
(312) 866-2553

For additional information circle #133

"THE MATCHBOX" ACTIVE INTERFACE AMPLIFIER FROM HENRY ENGINEERING

Designed to solve the common engineering headache of interfacing semi-pro or consumer equipment with professional studio systems, The Matchbox is a bi-directional, stereo device employing four independent amplifiers to provide simultaneous stereo inputs and outputs.

Two amplifiers convert a stereo high-impedance, unbalanced source to low-impedance, balanced outputs at studio operating level. A second pair of amplifiers converts a stereo balanced studio

New Products

line source to unbalanced outputs suitable for connection to the inputs of a semi-pro device.

All circuitry is active and direct-coupled for sonic transparency. The Matchbox contains an internal regulated power supply, and is small enough to allow it to be mounted to most cassette decks, VTRs, portable mixers, and similar equipment.

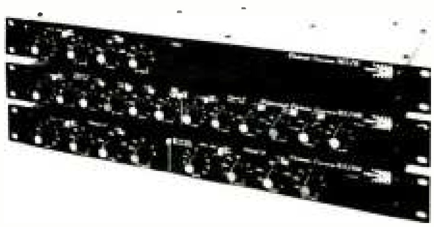
HENRY ENGINEERING
750 E. 5TH STREET UNIT 83
AZUSA, CA 91702
(213) 334-5580

For additional information circle #136

NEW FAMILY OF CROSSOVERS FROM BIAMP

The new series, which comprises the SX/35 five-way mono/three-way stereo, the SX/23 three-way mono/two-way stereo, and the MX/2, a mono version of the SX/23, feature 18 dB per octave slopes, and are switchable (normal or ×10) to expand and select the crossover points with greater accuracy.

All inputs and outputs are floating and balanced to maintain low noise, high slew rate, high level-output and complete isolation without transformers.



Other features include phase switching on HF outputs; input level control with +12 dB gain; subsonic filter (20 Hz); plus signal, power and peak indicator LEDs.

Frequency response is a quoted ±0.1 dB (20 Hz to 20 kHz), THD less than 0.01% (20 Hz to 20 kHz), and hum and noise, less than -87 dBm.

BIAMP SYSTEMS, INC.
P.O. BOX 728
BEAVERTON, OR 97075
(503) 641-6767

For additional information circle #137

EFFECTRON II SERIES OF DIGITAL EFFECTS DEVICES FROM DELTALAB

An enhanced Effectron series, upgraded features include increased flanging range, external infinite repeat, increased input range, stereo output, and lower prices. The Effectron II series consists of three models: the ADM-64, ADM-256 and the ADM-1024, with suggested retail prices of $299, $449 and $599, respectively.

The ADM-64 provides a full three octaves of flanging (8:1 flange ratio), and includes an internal envelope follower to provide additional flanging effects. Doubling and short echoes also can be achieved, with a delay range from 16 to 64 milliseconds.

The ADM-256 offers from 0.25 to 256 milliseconds of delay, while the ADM-1024 has a delay range of 0.25 to 1024 milliseconds. Both units will provide a range of effects, such as flanging, doubling, chorusing, and echo. By adding feedback and VCO, the following features also are possible: vibrato, tremolo, chorusing and multiple echoes. Each unit also includes an infinite (non-deteriorating) repeat button.

Unlike most other delay units, the Effectron II series units are said to maintain full audio bandwidth (16 kHz) and full dynamic range (90 dB typical) at all delay settings.

DELTALAB RESEARCH, INC.
19 ALPHA ROAD
CHELMSFORD, MA 01824
(617) 256-9034

For additional information circle #138

NEW SERIES 20 TWO-TRACK TAPE MACHINE FROM SOUNDCRAFT

The Soundcraft Magnetics Series 20 two-track master recorder, available with center-track option for timecode, is fully controlled by a microprocessor, and will automatically align itself for five different types of tape, and three EQ standards at any one of three speeds. All this information may be entered or altered by the operator and, once stored, the machine will align to the required settings at the touch of a button.

Standard features include a remote control unit, Dolby switching outputs, 14-inch reel capacity, varispeed, edit and dump modes, new membrane-type switches, and a headphone monitor socket.

SOUNDCRAFT ELECTRONICS
1517 20TH STREET
SANTA MONICA, CA 90404
(213) 453-4591

For additional information circle #139

AUDIO + DESIGN PROPAK INTERFACE UNIT WITH TIME CORRECTION FOR DIGITAL

As well as the recently launched Propak I unit for interfacing domestic and semi-pro equipment operating at

-10 dBV levels, with +4 dBm systems, the new Propak II with CTC™ (Coincident Time Correction) features +24 dBm headroom at inputs and outputs, and also is switchable for use with digital recorders operating to the EIAJ standard (eg. Sony PCM-F1) to produce a time-coincident digital stereo track.

Normally, with the EIAJ format inputs and outputs are multiplexed so that the recording has a time delay between channels. When reproduced through the D/A this is corrected. However, the possibility of taking a *direct* digital output means that tapes recorded this way are not PCM-1610 or Compact Disc compatible, because the signals are not time coincident. Propak II is said to provide a cost-effective solution, while maintaining playback compatibility with the Sony PCM-F1

Suggested price of the Propak I is $240; and the Propak II is $290.

AUDIO + DESIGN RECORDING
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #140

JBA-207 ISOLATION PRE-AMP FROM JBA RESEARCH

The new "Iso-balanced" pre-amp utilizes a discrete bipolar transistor design to provide a high degree of isolation, without the use of any transformer coupling. Designed specifically to interface any musical instrument with a mixing console, the iso-balanced design is said to surpass conventional balanced systems, providing up to 50 dB greater rejection of annoying ground-loop noise. This eliminates the need for any type of transformer coupling, thus eliminating transformer distortion.

The JBA-207 has two outputs: a high-impedance ¼-inch phone plug; and a low-impedance XLR "iso-balanced" output. The pre-amp may be powered by either a 9V battery, or by phantom power. Other features include a ground lift switch; input impedance greater than 1 megohm, handling a 3V peak-to-peak input; and THD less than 0.01%. Unity voltage gain is provided at the high-impedance output, and 10 dB of gain into a 600-ohm load at the low-impedance output.

JBA RESEARCH, LTD.
P.O. BOX 9632
MADISON, WI 53715
(608) 255-7100

For additional information circle #141

FURMAN ANNOUNCES LINE OF RACK-MOUNT AUDIO MIXERS

The new MM-4 is a low cost four-input mono mixer with effects send and

New Products

receive, and offers four high-impedance inputs suitable for use with a wide variety of signals. Gain adjusting channel level controls provide optimum signal-to-noise for a wide range of input levels. A switchable 100 Hz low-cut filter on each channel helps eliminate wind noise, low frequency stage noise, and compensates for the low-frequency boost of cardioid microphones used for close miking.

Effects send controls on each channel are mixed to a line-level effects output, while the effects return input is summed to the main output bus. Alternatively, the effects send output can be used to provide a separate mix for cue and fold-back. A high-output headphone amp,

with a separate level control, permits monitoring even in noisy environments. Both balanced and unbalanced outputs are provided, and can be used simultaneously.

The MM-4B is the same as the MM-4, except with the addition of low-impedance balanced inputs. Balanced and unbalanced inputs can be used simultaneously to mix up to eight signal sources.

The MM-8 is a stereo mixer with the same features as the MM-4, except for the addition of a second main summing bus and panpots to provide stereo (4×2) mixing; Model MM-8B is a stereo version of the MM-4B mixer.

FURMAN SOUND, INC.
30 RICH STREET
GREENBRAE, CA 94904
(415) 927-1225

For additional information circle #144

For additional information circle #146

STL TEST TAPES FOR 1500 SERIES INSTRUMENTS

A complete line of special test tapes and films for use with the Sound Technology 1500 Series test instruments — Models 1500A and 1510A — are available in all popular widths and speeds for reel-to-reel, broadcast cartridge, cassette and film formats.

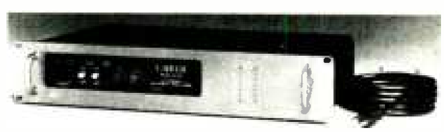
Used in conjunction with the Sound Technology system, these new tapes provide measurement of level, azimuth and frequency response of any tape or film transport. Flutter/speed test tapes also are available.

STANDARD TAPE LABORATORY
26120 EDEN LANDING ROAD #5
HAYWARD, CA 94545
(415) 786-3546

For additional information circle #145

CARVER UNVEILS 450-WATT MODEL PM-1.5 POWER AMP

Weighing just 21 pounds, and measuring only 19 inches wide, 3½ inches high, and 10¾ inches deep, the Carver Magnetic Field Power Amplifier PM-1.5 is described as a professional low-feedback, high-headroom amplifier capable of delivering 450 watts per channel.



Special features include fully proportional fan cooling; no "thermalling out"; recessed front panel controls; adjustable speaker protection circuit thresholds; remote turn-on sequencer with soft-start "power-up" mode; dynamic headroom controller; and reinforced front and rear rack mounts.

Power rating is a quoted 450 watts per channel into 8 ohms, 20 Hz to to 20 kHz, both channels driven, with less than 0.1% THD; less than 0.1% SMPTE IM distortion; frequency response -3 dB at 3 Hz and at 80 kHz; and damping of 200 at 1 kHz.

Suggested retail price of the PM-1.5 is $995.

CARVER CORPORATION
14304 N.E. 193RD PLACE
WOODINVILLE, WA 98072
(206) 483-1202

For additional information circle #147

FOSTEX DEBUTS X-15 MULTITRACKER FOUR-TRACK CASSETTE

The X-15 can record on up to two tracks at a time with individual tone and level controls; a built-in four-by-two mixer is used for monitoring during recording, and for setting pan and gain for each track during remix. The transport features soft touch controls, and tape format is compatible with the standard two-track Compact Cassette. Dolby-B noise reduction is integral to the unit.

"One of the most important features of the X-15 is that battery operation is standard," Fostex VP Mark Cohen explains. "Which makes it as handy and available for musicians and songwriters, as is the artist's sketch pad or reporter's notebook."

Retail price of the X-15 Multitracker is $495.

FOSTEX CORP. OF AMERICA
15431 BLACKBURN AVENUE
NORWALK, CA 90650
(213) 921-1112

For additional information circle #148

SYMETRIX MODEL 511 STEREO NOISE REDUCTION

According to the manufacturer, the 511 is a "non-complementary" or "single-ended" type system, which means that the unit does not rely upon an encode-decode process to accomplish the noise reduction function. An obvious advantage to the user is that the device may be used to remove existing noise from prerecorded tapes or, for that matter, from virtually any audio source, including noisy mixing boards, effects and processing devices, etc.

In operation, the 511 works to eliminate noise by incorporating a "soft-knee," voltage-controlled filter in series

. . . continued overleaf —

with a "soft-knee" downward expander. The user may select to use either circuit element independently, or both at once. The amount of actual noise reduction is controlled by a single variable threshold control, and visual indicators are provided for both filter and expander circuit operation.

The two channels may be used either as independent mono channels, or in the stereo link mode where exactly equal amounts of noise reduction will occur in both channels.

Manufacturer's suggested retail price for the unit is $595.00.

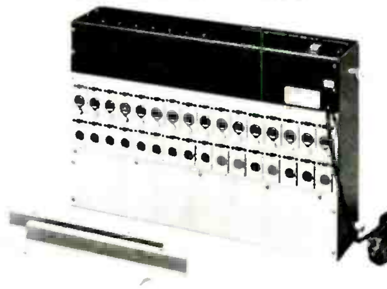
SYMETRIX, INC.
109 BELL STREET
SEATTLE, WA 98121
(206) 624-5012

For additional information circle #152

TASCAM LA-85 AND LA-40 LINE AMPLIFIERS

The LA-85 is designed to convert 85-16 and 85-16B multitracks to interface with +4 dBm balanced signals from three-wire, XLR-type connectors. The LA-40 matches balanced and unbalanced circuits, and serves as an adaptor between mismatched line levels or impedances. Both input and output circuits of the LA-85 are transformerless, and fully balanced. The balanced output stage of the self-powered LA-85 delivers 19.5V into a 600-ohm circuit that can drive long cable lines without suffering from HF signal loss.

The four-channel LA-40 permits interconnection between +4 dBm, -20 dBm and -10 dBV inputs and outputs, and also provides compatibility between Tascam or similar -10 dBV unbalanced inputs and outputs and almost all other audio equipment. Each of the four channels actually is two circuits back-to-back. By plugging a jumper across the RCA-in and -out jacks, the LA-40 isolates a line or converts from unbalanced to balanced operation when both the output and input are at the same +4 or -20 dBm nominal level. All connections are on the rear panel, and the phone jacks are on the front panel where they can be used in conjunction with a standard TRS jack patch bay.



Two individually switch-selected options are available for output operating level on the LA-85: +4 or +8 dBm. Output signal-to-noise is quoted at 100 dB, referenced to +4 with the DIN 20 Hz to 20 kHz filter, and crosstalk between sections 80 dB.

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

For additional information circle #153

ECHOTRON DIGITAL DELAY FROM DELTALAB FEATURES 4-SECOND CAPABILITY

Described as having been designed for customers wanting a solid-state digital delay loop with long delay capability, the delay range of the Echotron™ is from 256 to 4096 milliseconds, all at full 16 kHz bandwidth.

The unit also offers Infinite Repeat capabilities for storing sound digitally, without signal degradation. Sound-on-sound can be added by using the feedback control, in conjunction with the Infinite Repeat, allowing an engineer to produce over four seconds of repetitive audio.

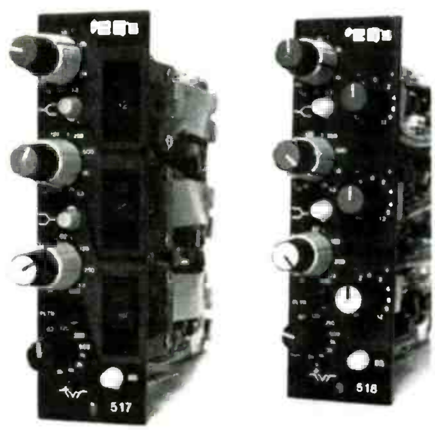
Manufacturer's suggested retail of the Echotron is $699.

DELTALAB RESEARCH, INC.
19 ALPHA ROAD
CHELMSFORD, MA 01824
(617) 256-9034

For additional information circle #154

MODULAR SPECTRAL SHAPING FILTERS FROM TROISI

Three fully parametric equalizer sections cover the audio spectrum, with enough overlap to allow concentrated control within each frequency band. Each section's bandwidth range is variable from 1/12 to over three octaves wide, with low- and high-frequency sections switchable from peaking to variably damped shelving response of 6 to 24 dB per octave.



EQ range is adjustable from infinite cut to 15 dB boost (Model 517 has 12 dB max boost) with center-off positions that remove the filter circuits from the internal signal path. A fourth-band, second-order filter is included to selectively remove unwanted portions of the audio spectrum. A continuously variable frequency control covers the nine octave range from 31 Hz to 16 kHz for either low-pass, high-pass or notch filter modes.

Both models come in a variety of package formats allowing for direct replacement of many 5¼-inch equalizer modules; Model 517 features thumbwheel cut/boost, while the otherwise identical Model 518 offers continuously variable controls. Unpackaged modules are available for OEM use and console retrofit, and custom packaging can be configured for special applications. Each equalizer module is priced below $400 in single quantities.

TROISI ENGINEERING & DESIGN CO.
27 RIVER STREET
WESTFORD, MA 01886
(617) 692-7768

For additional information circle #155

LOFT MODEL 400 QUAD LIMITER & NOISE GATE

The Model 400 is a combination four-channel, feed-forward limiter and noise gate whose front-panel controls include

noise gate threshold (minus-infinity to 0 dBv); limiter threshold (−12 to +12 dBv); and limiter attack/release time (1 millisecond to 1 second). A phase-reversal switch also is included on each channel.

A rack mount unit that occupies one, 1¾-inch rack space, the device comes standard with balanced inputs and ¼-inch phone connectors. Options include XLR connectors with electronically balanced inputs and outputs and/or recessed panel controls.

Suggested retail price of the Model 400 is $649.

PHOENIX AUDIO LABORATORY
91 ELM STREET
MANCHESTER, CT 06040
(203) 649-1199

For additional information circle #156

ELECTRO-VOICE UPGRADES EVM SPEAKER LINE

According to Greg Hockman, EV's

New Products

director of marketing/music products, the new EVM Pro-Line Series of loudspeakers offer increased power handling capability over existing Series II units, and are aimed at those wanting the ultimate in low-frequency reproduction in PA and sound reinforcement systems.

Several factors in the design, including heat-resistant materials, low-mass edgewound flat-wire voice coil construction, and proprietary manufacturing techniques, are said to account for the speakers' improved performance. The Pro-Line assemblies are driven by EV's largest 16-pound magnetic structure. Both the voice coil and the magnetic structure are vented to maximize heat dissipation in the voice coil area.

The new EVMs are available in three sizes and five models to fit virtually any design application. The 15- and 18-inch models can handle 400 watts continuous power (per EIA Standard RS-426A) and short-duration program peaks of up to 1,600 watts. The 12-inch EVM Pro-Line speakers are rated at 300 watts continuous, with 1,200-watt program peaks under the same test conditions.

The EVM-12L Pro-Line loudspeaker is intended for extended-range sound reinforcement applications. Almost identical to the EVM-12L, the EVM-12S is designed for more emphasis in the 2 to 3 kHz range for added brilliance and punch in full-range uses. The EVM-15B and EVM-15L are intended for general purpose, low-frequency reproduction, the latter for systems which also require response above 3.5 kHz. The EVM-18B Pro-Line offers enhanced low-frequency performance, including heavy fundamentals in the 30 to 40 Hz range.

The EVM Pro-Line characteristics are appropriate for both vented (bass reflex) and horn enclosures. Six optimally-vented enclosures have been designed by EV engineers specifically for the new speakers, and span low-frequency limits (3 dB down) from 38 to 83 Hz, and internal volumes from 1.3 to 13.0 cubic feet.

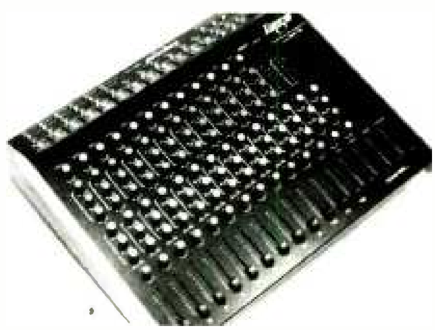
Pro-net prices of the new EVM Pro-Line are as follows: EVM-12S and -12L, $200; EVM-15B and -15L, $220; and EMV-18B, $330.

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

For additional information circle #159

TANGENT ANNOUNCES MODEL SX-12 LOW-COST STEREO MIXER

Three independent sends are provided in addition to the stereo mix. Adding a separate effects bus to the usual reverb and monitor busses enables an independent mix to be sent to a phaser, flanger or digital delay line. Or, with a simple modification, the extra effects bus can be converted into an extra monitor bus.



Standard features include 48-volt phantom power; three-band equalization on each input channel (100 Hz, 1.5 kHz, and 10 kHz); separate insertion jacks for patching external effects into individual channels; and a sloping rear panel for easy operation while in a road case.

Suggested retail price of the SX-12 is $1,200.

TANGENT
8206 NORTH DREAMY DRAW
PHOENIX, AZ 85020
(602) 997-4308

For additional information circle #160

AUDIO + DESIGN LAUNCHING RANGE OF AMBISONIC SURROUND-SOUND MODULES

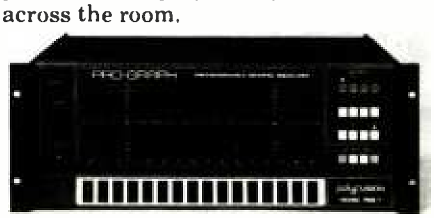
Designed to convert or transcode Ambisonic surround-sound information gathered from a Calrec Soundfield microphone system, or from conventional mono/stereo mikes for inclusion in four- or three-channel B-Format mixes, the new units include an Ambisonic Professional Decoder for decoding two-channel UHJ, and/or transcoding two stereo pairs or "quadraphonic" material to "quasi" surround sound, and for panning mono material in the soundfield; and an Ambisonic Multitrack Pan-Rotate unit that enables the positioning of a sound source anywhere in the horizontal or periphonic (three-dimensional) soundfield, and the rotation of one soundfield within another.

AUDIO + DESIGN RECORDING
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #161

MODEL PEQ-1 PROGRAMMABLE GRAPHIC EQUALIZER FROM POLYFUSION

The Model PEQ-1 Pro-Graph is a 16-band, programmable monophonic graphic capable of storing 64 response curves in its memory. Each curve is created or manipulated via step up/step down buttons under each frequency band. The variable intensity screen is comprised of a lighted graticule and a matrix of 240 LEDs, which are said to produce a display easily visible from across the room.



An A/B switch is provided to allow instantaneous comparison of the direct and equalized signals. The unit offers both "balance" and "unbalanced" inputs and outputs, while circuit technology ensures extremely low noise and distortion.

Pro-Graph comes equipped with two data input/output ports, allowing the unit to be remotely controlled by the optional PRC-1 Remcon unit, an automated mixing board, a computer, or other Pro-Graphs in a "daisy-chain" arrangement.

Frequency response is a quoted ±1 dB, 10 Hz to 20 kHz, signal-to-noise ratio 105 dB (unweighted), noise: -85 dBm, and THD: 0.0018, typical.

Boost/cut range is ±14 dB in 2 dB

New Products

increments at each center frequency (±7 dB in 1 dB increments as option), at 16 frequencies between 16 Hz and 16 kHz, on 2/3-octave spacing.

Price of the Model PEQ-1 is $1,695.

POLYFUSION ELECTRONICS
92 BENBRO DRIVE
BUFFALO, NY 14225
(716) 681-3040

For additional information circle #165

CANARE INTRODUCES BROAD LINE OF CABLE AND CABLE SYSTEMS

Of special interest is Canare's L4E Series, of four-conductor cable that consists of two twisted pairs, plus a high-density braided shield. Because each "conductor" in the balanced cable actually consists of a twisted pair, the included area between conductors is minimized which, in turn, maximizes rejection of AC hum and all forms of electro-magnetically induced noise. Tests show the resulting noise immunity is 10 times better than standard balanced mike cables, Canare claims.

The company's cable reels are available for single or large multipair cables, with or without connector panels. Several models come with three-position brakes that regulate tension or lock the reel completely. Most reels can be stacked for storage and easier pulling of long lines, and some models have roll around casters.

A variety of low-crosstalk, multipair cables for construction of custom snakes also are available, including factory-wired, 8- to 32-channel snakes, with multipin connectors that mate either to junction boxes, or to XLR pigtails. Special combinations of optimized audio, video and communications cable in one multiconductor bundle come in five common remote truck formats.

CANARE CABLE, INC.
6733 VINELAND AVENUE
NORTH HOLLYWOOD, CA 91606
(213) 506-7602

For additional information circle #166

SOUNDER MODEL 500 PHASE CHECKER

The new unit determines and displays the polarity of all audio equipment, and quickly and easily eliminates the problems of phase distortion, bass cancellation, and loss of acoustic power caused by polarity errors.

The battery-powered Checker consists of pulse generator to excite the system, and a polarity detector which shows the results. An internal electret condenser mike in the detector, and XLR, ¼-inch and RCA phono jacks on both units enable testing of equipment.

SOUNDER ELECTRONICS, INC.
21 MADRONA STREET
MILL VALLEY, CA 94941
(415) 383-5811

For additional information circle #167

DELTALAB SUPER TIMELINE SERIES DIGITAL SIGNAL PROCESSORS

Comprising full bandwidth (16 kHz) programmable digital delay signal processors, both the ADM-512 and the ADM-2048 are described as being simple to program, and easily addressable via the front panel or the accessory remote control.

The ADM-512, with a suggested retail price of $799, features flanging, doubling, chorusing and echo effects, with up to 512 milliseconds of delay.

The ADM-2048 is identical to the ADM-512, except that it provides the user with over two seconds (2048 milliseconds) of delay; the ADM-2048 has a suggested retail price of $999.00.

DELTALAB RESEARCH, INC.
19 ALPHA ROAD
CHELMSFORD, MA 01824
(617) 256-9034

For additional information circle #168

ELECTRO-VOICE FR15-2 TWO-WAY SOUND REINFORCEMENT SPEAKER

Intended for sound reinforcement applications where a wide, controlled coverage angle and high efficiency are desired from a single cabinet design, low-frequencies of the FR15-2 are handled by a 15-inch EVM-15L Series II woofer mounted in a 4.3-cubic foot optimally vented enclosure. Frequencies above the 1.5 kHz crossover point are handled by a compression driver mounted on a 90- by 40-degree, constant-directivity horn. Unlike conventional horns, EV's constant-directivity horn is said to maintain its rated beamwidth to the highest frequencies, assuring broad, uniform coverage without dead spots.

The frequency response of the FR15-2 is described as being essentially flat from 50 Hz to 15 kHz. The speaker's sensitivity is rated at 98 dB SPL with a 1 watt input, measured at 1 meter on-axis. The long-term power-handling capacity is 200 watts, measured using shaped pink noise with a 6-dB crest factor.

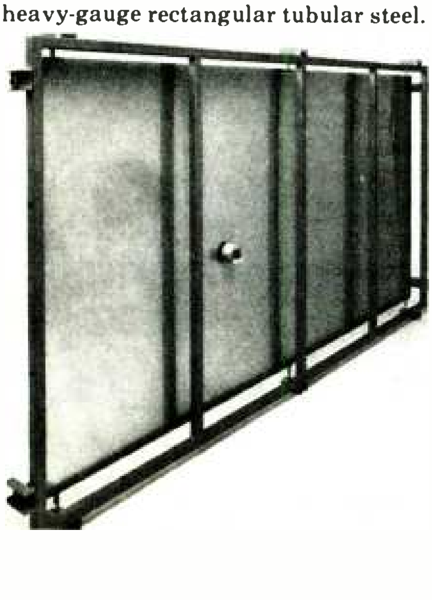
Weighing 94 pounds and measuring 28⅜ by 31½ by 16⅝ inches (H × W × D), the new FR15-2 speaker has a pro user net price of $665.

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

For additional information circle #169

LOW-COST MODULAR REVERB SYSTEM FROM REASONABLE ALTERNATIVES

Described as the first totally modular plate reverberation system, the heart of the new "Blue Plate Special" is a full-sized (3- by 6-foot) selected, stainless steel sheet, with spot-welded, double-thick, steel reinforcements at all suspension points. Mounting hardware consists of allen bolts and cast iron yokes, supported by rubber and steel washers. The frame is welded from heavy-gauge rectangular tubular steel.



All that is required from the user is a cue system with amplifier, and one or two contact mike/pickups. Other modules offered include pickup/preamp with variable EQ, damping mechanism, high-power drive amp with EQ, and an enclosure with integral suspension/shock mount.

Suggested list price of the Blue Plate Special is less than $700; it can be seen in booth #118 at the forthcoming New York AES Convention.

REASONABLE ALTERNATIVES
P.O. BOX 733
CRANFORD, NJ 07016

For additional information circle #170

MARSHALL MEMORY MODULATOR DIGITAL EFFECTS PROCESSOR

Due to be introduced at the forthcoming AES Convention in New York, the Memory Modulator is a computer-managed, programmable delay processor that is said to depart radically from normal devices in this catagory. The company feels that the area of human interface in programmable devices has long been ignored, or poorly planned. All other devices have a front panel that attempts to emulate older physical or analog panels. This becomes meaningless as soon as the first program is recalled, since the values stored have nothing to do with the values shown on the panel.

The unit has no conventional controls at all; instead, a large 20-character alpha/numeric display shows the exact values of all parameters on request, in normal English. The user can call up instantly and directly edit any program

New Products

in residence, and the device will lock to tape and track effects originally used (it will even load those effects off the multitrack into another Memory Modulator, and employ them). It can actually lock to the data track, and repeat dynamic effects as well (hand-turned control changes during the piece). Multiple units may be locked in any of several arrangements and driven from one data control track, including loading the actual programs into the units.

The swept delay range is over 2000:1; modulation waveforms include automatic envelope following, log and linear LFOs, and all parameters, including EQ, are programmable.

MARSHALL ELECTRONIC
1205 YORK ROAD, SUITE 14
LUTHERVILLE, MD 21093
(301) 484-2220

For additional information circle #173

SOUNDCRAFT UNVEILS TS24 IN-LINE CONSOLE

Marking the beginning of a new range using in-line technology to supplement the company's range of split consoles, the new TS24 is said to feature a new approach to in-line working, and is designed to greatly simplify operation and dispense with the confusion that can occur when using current layouts.

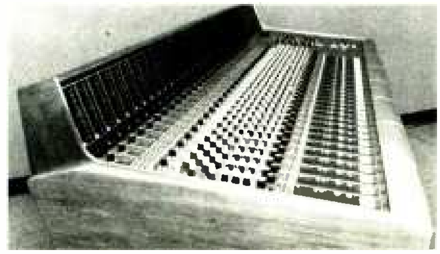
With the use of a simple system comprising a few master controls, the status of the whole console can be changed quickly from one mode to another. Monitoring facilities during record and overdub also have been greatly simplified, and the console electronics are of the usual high standard expected from the manufacturer. The TS24 is described as providing a great many facilities, at a very reasonable price.

SOUNDCRAFT ELECTRONICS
1517 20TH STREET
SANTA MONICA, CA 90404
(213) 453-4591

For additional information circle #175

SOUND WORKSHOP LAUNCHES SERIES 34 MIXING CONSOLE

The new Series 34 features 24 mix and six auxiliary send busses, four-band sweepable EQ, monitor EQ insert, dual stereo echo returns, and optional computer-controlled VCA package, featuring ARMS automation.



A dual metering system enables simultaneous display of both input signal chain, and the output (bus/tape) signal chain. The bus/tape meter is a three-color, 40-segment high resolution design with 40 dB dynamic range, peak and average ballistics, and variable intensity.

All main inputs and outputs feature balanced transformerless design, with output circuit clip level exceeding +27 dBu. The optional computer/VCA package features in-place solo, Super-Group input subgrouping, and tape-based console automation with independent mute write.

The Series 34 standard configuration is 28-in/24-out in a 32-input mainframe. Pricing is at $31,400 for the standard configuration, and $39,900 for the computer/VCA package.

SOUND WORKSHOP
1324 MOTOR PARKWAY
HAUPPAUGE, NY 11788
(516) 582-6210

For additional information circle #174

CETEC VEGA R-42 PRO PLUS WIRELESS-MIKE RECEIVER

The new Model R-42 features "infinite gain" technology, ultralow noise, true dual-receiver diversity, and switch-selectable Dynex II. System dynamic range with Dynex II is said to be typically 108 dB, A-weighted (maximum deviation to noise floor). Even with Dynex II switched out, the receiver has a 92-dB (A-weighted) signal-to-noise ratio.

Highest adjacent-channel rejection is achieved with 16 poles of IF filtering. The preselector is a true four-pole helical-resonator filter (silver-plated). Infinite gain technology is said to ensure the best possible signal-to-noise ratio at low signal levels, and improved processing of multipath RF signals.

CETEC VEGA
9900 BALDWIN PLACE
EL MONTE, CA 92667
(213) 442-0782

For additional information circle #179

QUANTEC DIGITAL ROOM SIMULATOR NOW AVAILABLE IN U.S. FROM MARSHALL

Several important changes will be immediately put into effect as a result of the recent announcement. First, the new price of the Room Simulator will be reduced by a full 20% from $12,500 to $10,000. Second, all units will be shipped from stock at the Marshall factory, eliminating the long post order wait that existed before. Third, the dealer network will be increased dramatically.

The Quantec Room Simulator is a linear 16-bit digital ambience and reverb processor whose algorithms are said to be consistent with actual acoustics of real rooms. Room parameters are defined in cubic meters, and can range from rooms smaller than (and inserted into) your head, to giant concert halls with reverb times of up to 100 seconds at 1 kHz, and multiples of that at the high- and low-frequency limits.

The QRS is a true stereo-in, four-channel-out device, with the inputs configured as two "speakers" in a room, and the outputs as four "microphones" placed in that room. The unit is said to exhibit a true non-pumping (no noise reduction schemes of any type are used) 94 dB A-weighted dynamic range, with very special guard filters designed to eliminate that "closed" or "tight" feeling produced by normal filters. Reverberation density exceeds 10,000 per second. Infinite reverb times and special non-reverb room simulations also are included.

MARSHALL ELECTRONIC
1205 YORK ROAD, SUITE 14
LUTHERVILLE, MD 21093
(301) 484-2220

For additional information circle #176

AUDIO + DESIGN TCR1 PORTABLE TIMECODE READER

The TCR1 SMPTE/EBU timecode reader and regenerator is said to meet the need for a simple, economic and accurate code-reader. The unit operates for up to 2,000 hours from one set of internal batteries, and will read timecode from serial code outputs.

Apart from reading SMPTE/EBU code, at the flick of a switch the unit will display user-bit code. Drop-frame and color frame are indicated below the frame-count digits. The TCR1 also has an output through which timecode can be accessed "cleaned-up."

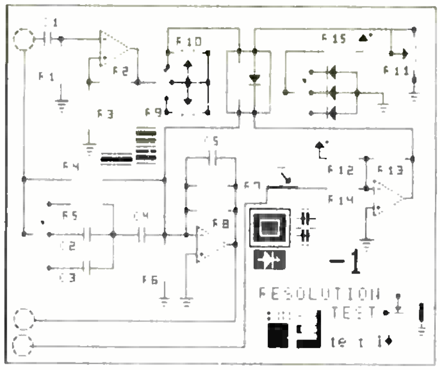
AUDIO + DESIGN RECORDING
P.O. BOX 786
BREMERTON, WA 98310
(206) 275-5009

For additional information circle #177

MARSHALL ELECTRONIC LAUNCHES SAINTSOFT COMPUTER DRAFTING SYSTEM

The new Saintsoft CADDRAFT computer-aided system is a low cost, ultra high speed drafting tool that is said to differ from conventional CAD systems in several ways. Designed to respond and operate typically 20 to 40 times faster than conventional CADs, moving or editing sections of a drawing is done visually, immediately, and physically without the need to define target or displacement vectors.

Everything the user needs to start drawing is included; for example, in the schematic mode over 100 schematic objects are supplied prescaled and grid referenced. These can be modified, combined, stripped, or edited in any way,



and stored as additional objects. A module is also provided for the generation of custom shapes.

CADDRAFT incorporates a 12-screen, real-time buffer and a high-speed program buffer. It offers features such as removable grids, grid snap, step and repeat, microscopic editing, auto measuring, auto pattern fill, estimation, auto parts labeling and sequential numbering, instant multi-screen pan — and all with no need for redraws.

The CADDRAFT system will be on display in booth #1440 at the forthcoming AES Convention, New York City.

MARSHALL ELECTONIC
1205 YORK ROAD, SUITE 14
LUTHERVILLE, MD 21093
(301) 484-2220

For additional information circle #178

EQUIPMENT UPDATE

— *continued from page 26* . . .

with.

The accompanying illustration shows the compander after its changes. The output of the second-order linear-prediction filter is a "locally decoded" version of the digital bit stream, since the delta-mod decoder reconstructs the audio by driving the same bit stream into this same loop filter. Therefore, by deriving the VCA-gain information from the loop-filter output in both encode and decode, the VCA gains are guaranteed to be complementary; i.e., mistracking cannot occur.

Two VCAs are used in both the encoder and the decoder to provide the spectral and the amplitude compression. The amplitude-control VCA merely adjusts the broadband gain of the signal, whereas the spectral-compression VCA acts as a voltage-controllable filter (VCF) with a constant "hinge" frequency. The filter is capable of providing up to 30 dB of boost or cut at 20 kHz.

During encode, the high frequencies present at the input are dynamically sensed by the spectral-control RMS detector. The output of this detector is used to control the VCF response such that when strong high frequencies are present they are attenuated, and when only weak high frequencies are present they are boosted.

During decode, the control voltages to the expander VCA and VCF are inverted in polarity from those of the compressor. All of the VCAs obey an exponential control law, so that inverting the control voltage results in reciprocal VCA gains. Moreover, the voltage-controlled filter is symmetrical in design, which means that inverting the gain also inverts the frequency response.

A secondary, "transient-speedup" circuit is employed in the level-sensing path to allow faster gain reduction during input transients. It works by looking for strings of "1s" or "0s" in the digital output, which indicate the onset of slew-rate overload in the modulator. If more than about 20 consecutive 1s or 0s are found, a DC voltage is applied to the RMS-detector input. This voltage is about 10 dB higher in level than the largest AC input signal normally present at the detector input, which causes the detector to change its output 10 times faster than normal. This ensures that transients are not clipped in the decoded output.

The production 700's final dynamic range, then, is greater than 110 dB at low and middle frequencies, and only slightly less than 110 dB at high frequencies. This means that the unit's capabilities are quite close to those of the human ear. Note that the noise measurement in this dynamic range figure is unweighted, and for the full 20 Hz to 20 kHz bandwidth. Also important in any companded design is the signal-to-noise ratio in the presence of a signal. In this area, too, the 700 delivers very high performance, having around 85 dB at 1 kHz. This is due to the action of the spectral compressor, which significantly reduces high-frequency noise in the presence of low frequencies.

One nice consequence — the creative edge, if you will — of having a full-bandwidth dynamic range of around 100 dB is that even if as much as 20 dB is lost or sacrificed somewhere in the recording and mastering processes, the result will still be a master with a dynamic range as wide as the Compact Disc can handle (90 dB or so).

Field test also revealed that crude editing splices (made without a video editor) would not be fully corrected, but could produce a very small (two-millisecond) click. This has been changed so that now a gross cut produces brief muting — which will make edits made during silences (assembly editing, for example) quiet and seamless.

Delay Line for Analog-Disk Mastering

We have produced a working prototype of a delay unit for disk mastering, to be called the D700 Disk-Mastering Delay, adjustable up to two seconds. Ours differs from some other digital delays in that there is no degrading whatsoever of either the delay or the original in the delaying process. The delayed signal is what gets sent to the cutting head, while the original 700 signal is used for previewing the program, and controlling the cutter spacing with total accuracy.

Video Synchronization

We thought it would be useful for studios to be able to link several 700s together for multichannel recordings, or to synchronize a 700 to the house video-production system. Production 700s, then, have video-sync inputs (and outputs as well).

With varying degrees of activity, dbx is at work on developing open-reel, fixed-head, razor-editable two- and multitrack machines, competitively priced with analog recorders of similar format; full digital editing for same; transcoding (going from CPDM directly to PCM for the production of CDs); and consumer units.

Although parts and other costs have risen in the last year, we have succeeded through a number of means in coming in under our projected figure of "less than $5,000." ■■■

news

MITSUBISHI ELECTRIC ACQUIRES DIGITAL ENTERTAINMENT CORP; ADDS SUPPORT STAFF

In making the recent announcement, Yoshito "Super" Yamaguchi, chairman of Mitsubishi Electric said, "We see the professional audio and video technologies as ones of tremendous growth in the coming years, and have decided that the time is now ripe to increase our investments in these areas. The success of the new Compact Disc technology has opened the doors for increased activity in the entertainment areas, and this new venture will very quickly find itself uniquely suited to developing new and exciting ways to market our sophisticated technologies."

Along with Mitsubishi's digital audio products, DEC also will develop and market a new system of interactive digital audio storage devices, music software manufacturing equipment, entertainment-related business computer systems, and other related pro-audio products.

Yamaguchi will serve as DEC's chairman, while Tore B. Nordahl will remain as the president and chief executive officer. Immediate plans include the staffing of offices in New York, Los Angeles, and Nashville, where regional sales and technical support services will be maintained. Sonny Kawakami of Mitsubishi is assuming the position of VP Marketing for DEC, while Lou Dollenger (Mitsubishi Electric in Chicago) is moving to the New York area to become marketing manager. Bill Van Doren is regional manager at the DEC Hollywood office. Van Doren comes to DEC with a nine-year sales career with

news:

Studer-Revox America, where he was instrumental in the growth and acceptance of Studer products in the U.S., particularly the company's multitrack recorders.

TECHNIGROUP MARKETING AND SALES COMPANY FORMED BY CHRIS COFFIN

Coffin, formerly with Sound Workshop, says that TechniGroup, Inc. will serve as a national marketing, sales and support organization for OEMs in the audio/electronics industry. The new corporation will be based in Northport, NY, with a California operation to be announced in the near future. A nationwide computer communications system will be an integral part of TechniGroup's operating and support structure.

TechniGroup and Marshall Electronic have joined forces to increase marketing and distribution of Marshall's newly expanded product line, including the Memory Modulator, The Quantec Room Simulator, and Saintsoft CADDRAFT computer-aided drafting package.

For more details, contact Chris Coffin at: TechniGroup, 16 Green Acre Lane, Northport, NY 11768. (516) 261-5541.

STUDER SELECTS FIRST U.S. DEALERS

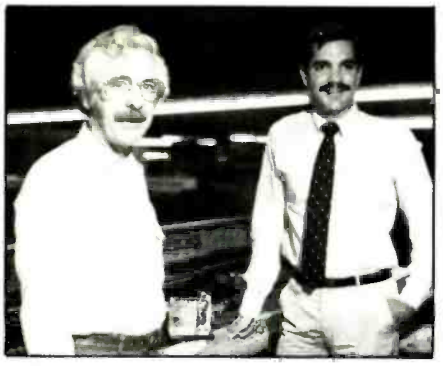
The company's equipment now may be purchased through a limited network of independent professional audio dealerships; previously, all products in the Studer line were made available exclusively through direct distribution by Studer Revox America, Inc.

Dealers currently authorized to carry the Studer line are: Audio Engineering Associates of Pasadena, CA; Bridgewater Custom Sound of Harvey, IL; Doug Brown Enterprises of Tulsa, OK; Cramer Video of Needham, MA; Midcom, Inc. of Arlington, TX; Emco, Inc. of Rockville, MD; Pro Audio General Store of Atlanta, GA, Coral Springs, GL, and Carol Stream, IL; and Studio sonics Corporation of Schaumburg, IL.

Studer mixing consoles, A800 multitracks, and A80VU stereo and 16/24-track machines will continue to be available only through direct distribution by Studer Revox America.

AWARD WINNING QUAD-EIGHT FILM CONSOLE DESTINED FOR AUSTRALIA

Colorfilm Pty., Ltd. of Australia has purchased from The Burbank Studios the original Quad-Eight Dubbing 5 custom rerecording console for which the manufacturer won an Academy of Motion Picture Arts & Sciences Technical Award in 1974. Using the console, TBS post-production department also received many Academy Award nomi-



Colorfilm sound director Les McKensie (left) with Quad-Eight president Cam Davis.

nations, including *Electric Horseman* and *Tootsie*, and an Oscar for *All The President's Men*.

In June, Quad-Eight delivered to TBS a new 24-foot, 108-input, six-track stereo console, which presently is being used in Dubbing 5. But since the original board was far from ready to retire, Colorfilm contracted with Quad-Eight to make some minor modifications, and recheck the entire console for original performance specifications. The like-new console then will fulfill the requirements for Dolby Stereo work in Colorfilm's main theater, which is equipped with 23 RCA high-speed film transports, and a Studer A800 multitrack.

ALTEC LANSING MOVES CORPORATE HEADQUARTERS

Following the sale of its 14-acre Anaheim, California, facility, Altec Lansing has relocated its corporate offices to another facility, also in Anaheim, that will house Altec administration, engineering, sales and marketing departments. All Altec manufacturing operations now have been consolidated at the company's Oklahoma City facility.

The new address for Altec corporate headquarters is: Altec Lansing, 1250 Red Gum Avenue, Anaheim, CA 92806. (714) 632-7717.

3M SELLING SPARE PARTS INVENTORY FOR ANALOG RECORDERS TO ELECTRO-TECHNOLOGY CORP.

The sale of service support capabilities and spare parts inventory to Electro-Technology Corporation, Menlo Park, California, includes a licensing agreement to manufacture spare parts required to repair or rebuild the analog recorders that were last manufactured in 1979 by 3M's former Mincom Products Division.

According to Art Cuscaden, technical service supervisor of 3M's Broadcast and Related Products Division, the agreement includes all existing spare parts, engineering data, vendor information and test and manufacturing fixtures needed to provide repair services or parts to current owners of the equipment. In addition, the agreement provides for the training of Electro-

NEWS *continues on page 222* —

For additional information circle #186

Tangent & The Extra Mix You Always Wanted.

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For additional information circle #187

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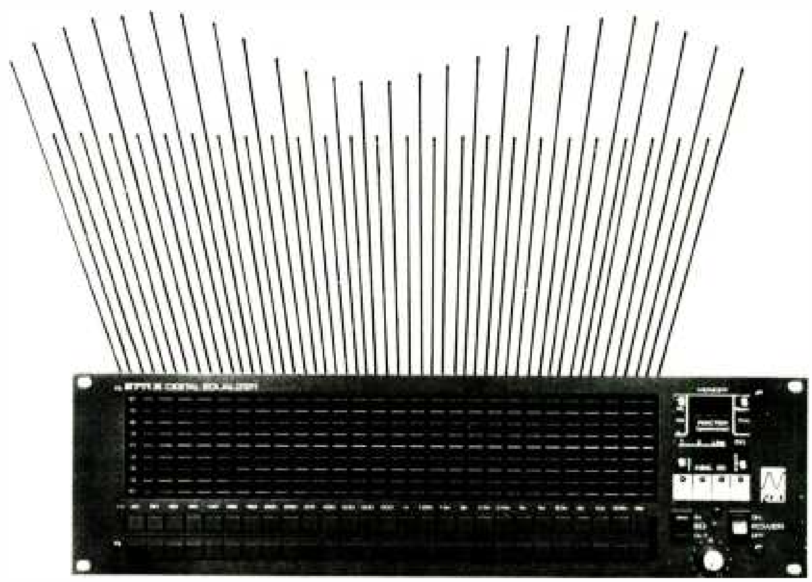

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For additional information circle #190

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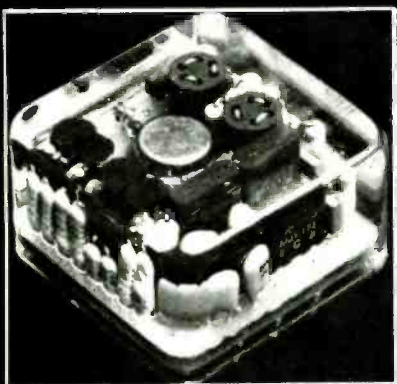

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For additional information circle #191

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Attn: Jim Yamaguchi
Manager, Technical Services

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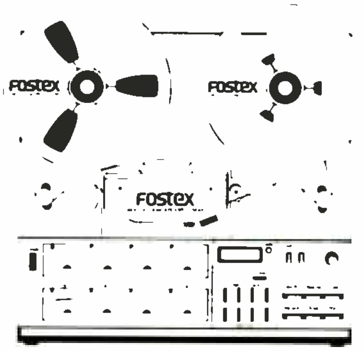

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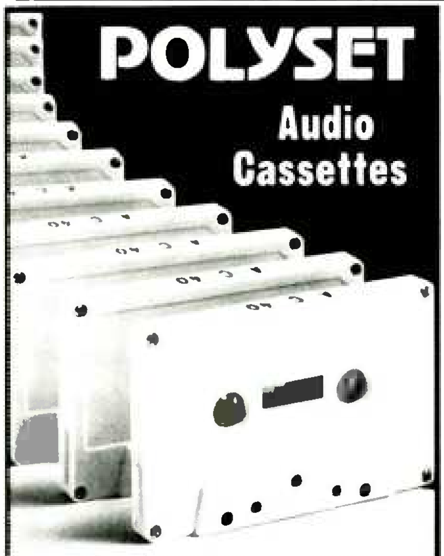

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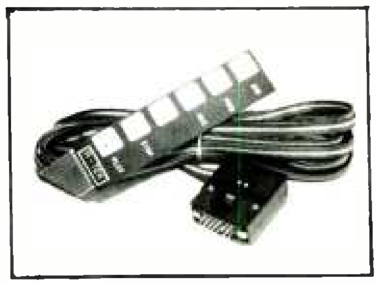

20 foot ribbon cable interface connector allows record, play, fast forward, rewind, stop, and pause functions *without* the need for a bulky remote box.

The RS-1 Remote Control Strip, which measures 5½ by 1½ inches, has an adhesive backing for mounting on any flat surface, such as a blank mixer module.

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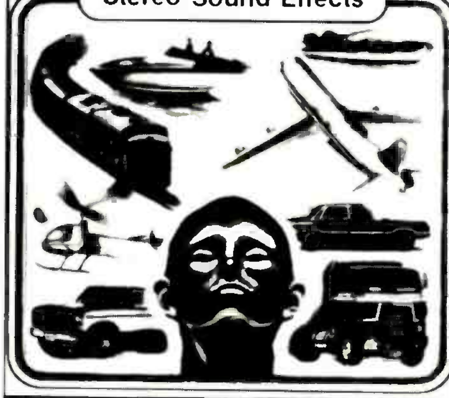

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news

— continued from page 215 . . .

Technology personnel in the use of the fixtures and equipment.

"We are pleased to announce the agreement with Electro-Technologies," Cuscaden says. "The company has been in business since 1972 and has an excellent background in systems design, electronics and mechanics that will assure continued support to our analog audio customers." However, the recent announcement has no effect on 3M's continued efforts and commitment to the field of digital recording technology.

EMIL HANDKE JOINS VALLEY AUDIO

According to Bob Todrank, president of Valley Audio, Emil Handke has been hired at Valley Audio as general manager from his present position as national sales manager for Sound Workshop, Inc.

"I'm very excited about having Emil join us at Valley Audio," Todrank says, "and feel he will make a major contribution to the continuing growth and success of Valley Audio. His background in all facets of the professional audio industry plus his friendly personality make him especially suited for his role in Valley's future.

"Emil's presence here will give me the opportunity to concentrate my efforts toward consulting and large system sales, which are the two areas I enjoy most. Our plans for Valley Audio over the next three years include a great deal of expansion and diversification. Thus it was necessary to bring on board top level management people during the initial stages."

Lexicon's chief executive officer, Ron Noonan (left), not only runs one of the world's leading pro-audio and broadcast equipment companies, he also has joined a select group of captains to win the coveted Marion cruising race. Sailing in G class with his daughter Michelle, son Rick, and crew, Noonan's Bristol 40 sloop *Wildflower* won overall first place, competing against 134 yachts in the 645-nautical-mile race to Bermuda.

AUDIO KINETICS MERGE WITH MELKUIST; UNVEILS MASTERMIX AUTOMATION SYSTEM

As a result of the recent merger, Audio Kinetics now offers the Mastermix Console Automation System. Based on a 5¼-inch floppy-disk storage system, and using Melkuist GT800 technology, the new package is described as the most compact, cost effective, SMPTE-based system yet developed. The advantage of using disk for data storage is that only one track on the master multitrack is used for SMPTE timecode. The system offers automatic updating with frame-accurate timing precision.

Mastermix has an integral multistandard timecode generator, capable of reverse code generation, for tapes coded in error from the start of the take, and subsequently need pre-roll information. A simple keyboard provides the option of manual or automatic control. With 600K bytes of disk storage, Mastermix has typically 30 minutes of mix duration, in each of four memories, plus a 64K byte RAM scratchpad memory. A serial data port (RS232/422) is provided for communication with Q.Lock synchronizers and other computer peripherals. The system will interface to consoles with either processor based or DC grouping systems. Other features include the ability to automate fader level and mute functions independently, automatic edit and merge, and fast back-up of complete mixes onto separate disks. ■■■

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Photo courtesy New River Studios, Fort Lauderdale, Florida

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