RECORDING
ENGINEER/PRODUCER

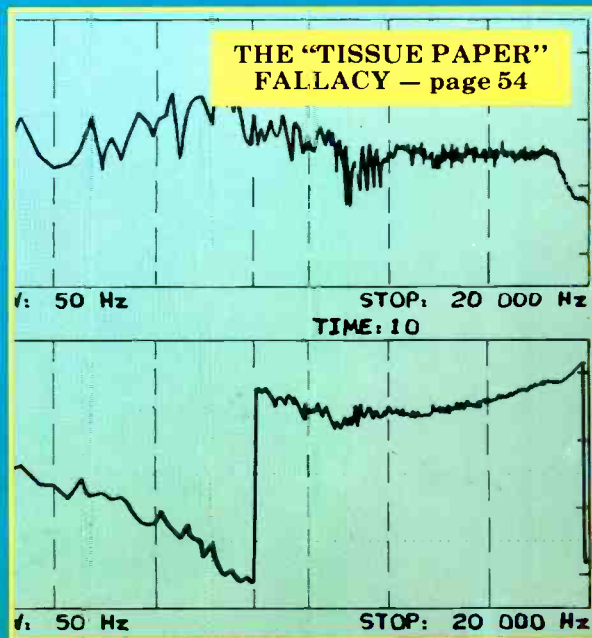PRODUCING AUDIO FOR ● TAPE ● RECORDS ● FILM ● LIVE PERFORMANCE ● VIDEO & BROADCAST
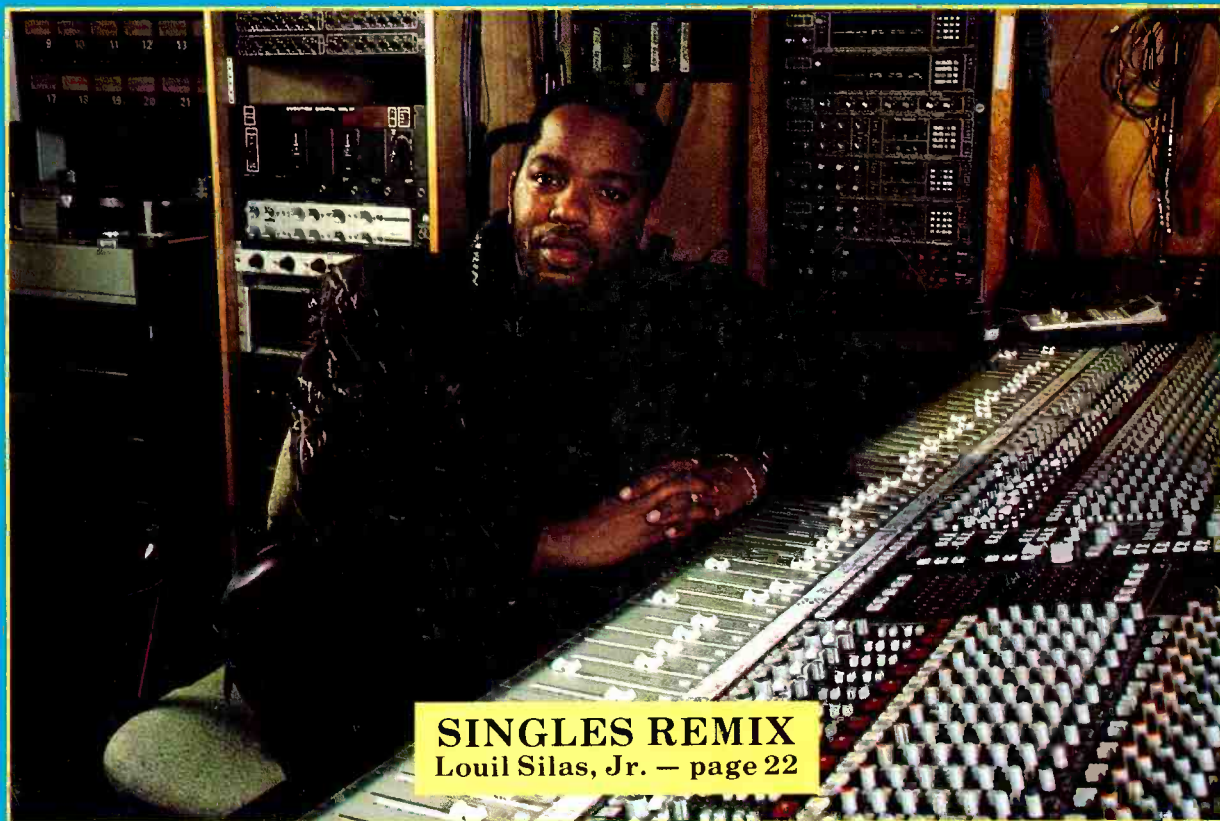
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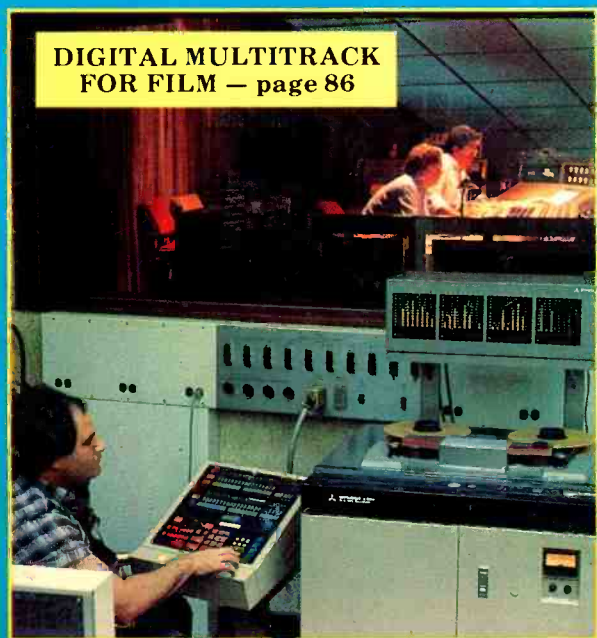
DIGITAL MULTITRACK FOR FILM — page 86

THE "TISSUE PAPER" FALLACY — page 54

V: 50 Hz   STOP: 20 000 Hz
TIME: 10

V: 50 Hz   STOP: 20 000 Hz

SUPERSTAR
by quad eight

Advanced technology and unparalleled flexibility come together in the SUPERSTAR music recording console. Development of this console centered around the dual requirement of truly high definition sound and low noise, so critical for digital recording.

No other single console offers the combination of superior sound and flexibility in size and layout at such an affordable price. Field expandable, the SUPERSTAR provides ergonomical positioning of the console modules, allowing you to satisfy your own configuration needs. High resolution meters, central bus assignment, Intelligent Digital Faders, and the most comprehensive automation system all add up to SUPERSTAR – your next console.

MODULAR CONSOLE

The SUPERSTAR is a totally modular console using air frame design concepts for strength and rigidity. Individual frame sections are in groups of 8 modules, with plug in wiring for true field expandability. The modular overbridge accepts the new limiter/compressor/gate for use either in-line with the input module or as a peripheral.

60-segment LED bargraph meters use advanced circuitry for precise and stable indication, offering VU, Peak, VCA level, and Spectrum Analyzer displays switch selectable.

Plug-in interchangeable equalizers and preamplifiers in each I/O module give instant user selectability and allow the addition of new technology at any time. Each module is of dual-purpose in-line design with line trim, equalizer, filter, 8 echo/cue sends, and fader switchable into the monitor/mixdown or main channel. Monitor/mixdown can be assigned to two independent stereo output busses for added versatility.

CENTRAL ASSIGNMENT

This electronic output assignment cross-point switching system assures fast and reliable connections from the console to your tape machines with full routing or mixing capability. 64 output busses are assigned from each input module by a central touch control plasma display panel controlling up to a 96 by 64 electronic switching matrix. Completely software driven, the panel allows instant selection and display of the bus assignment with 10 presets in local memory. Optional unlimited storage to disk is provided. Easy to use, the system prompts for bus assignments and provides help through informative menu displays.

The building block matrix system consists of 16 by 16 switching cards bussed to 16-output summing cards. Logic controlled monolithic switching elements use zero volt current switching for extremely low distortion and feed through.

THE NEXT GENERATION

The introduction of the SUPERSTAR signals a new era in professional sound control. With more and more studio facilities acquiring digital multitrack recording capabilities up to 64-track, larger sophisticated console systems with transparent sound performance are necessary. Digital signal processing (DSP) is neither economically feasible nor technologically advantageous today. A new generation analog console with advanced digital control is required to bridge the gap between the DSP consoles of the 1990s and the currently marketed analog consoles of the 1970s. The SUPERSTAR is such a console system. See it before you decide.

INTERTEC PURCHASES *RECORDING ENGINEER/PRODUCER*

On January 17, Intertec Publishing Corporation, through its parent company, MacMillan Publishing Corporation, finalized its acquisition of *Recording Engineer/Producer*. The announcement was made by Jack Hancock, Intertec President, and Cameron Bishop, Group Vice President. Formerly owned and published bi-monthly by Gallay Communications, Inc., *R-e/p* will be published by Intertec beginning with this, the February issue.

The magazine's editorial offices, headed by Mel Lambert, Editor, will remain in their present Hollywood location. However, all other publishing functions will be performed from Intertec's home office in Overland Park, Kansas, including business management, advertising coordination, marketing, sales promotion and production of the magazine.

Intertec's other immediate plans for *R-e/p* include a monthly publishing schedule to begin in June; field sales offices in New York, Kansas City and Los Angeles; a complete graphic redesign; and refined sales and advertiser support materials. Circulation plans include application for BPA (Business Publication Audit of Circulation) audit within the next 12 months.

Recording Engineer/Producer makes the ninth electronic communications trade magazine published by Intertec Publishing Corporation. Other Intertec magazines include *Broadcast Engineering, Video Systems, Radio y Television, Sound & Video Contractor, Electronic Servicing & Technology, Microservice Management, Land Mobile Product News,* and *Cellular Business*.

From the Editor:

As we enter the second half of the Eighties, a strong sense of enthusiasm and optimism permeates the Pro-Audio Industry. In fact, many would consider that our industry has never been in better shape, and is now facing a very healthy future. The recent upswing in business has been shared by recording studios, broadcast production, audio-for-video, film re-recording, and concert-sound facilities alike. This positive trend can be traced to several key factors:

• The increasing application of digital recording in just about every type of audio session, a move that has been fueled, in large part, by the unprecedented success of the consumer Compact Disc.

• In addition, the growing importance of high-quality audio-for-video, both for network and independent Stereo TV productions, as well as consumer release on VHS and Beta HiFi videocassette.

• A trend towards electronic-music production, and the routine use of computer-based MIDI systems in the control room. While many such scoring and compositional projects can be started in a smaller, less well-equipped facility, there is no getting away from the fact that few individuals can afford to provide themselves with the virtual arsenal of sound processors available at commercial studios.

And, since this is the first issue of a new year, I'd like to offer the following predictions of the ways in which our industry might be shaped over the next 12 months:

• Digital technology will continue to penetrate all sectors of the recording and production industries, in particular the development of random-access editing and mixing systems. Products such as the AMS AudioFile, Compusonics DSP Series, DroidWorks SoundDroid and prototype Lexicon Project RD-1, will be augmented in the near future by systems from several individuals and companies I know to be actively working in this area. As the cost of random-access memory, high-speed 32-bit microprocessors and hard-disk drives continues to fall, and front-end software becomes tailored to meet the specific needs of working professionals, I predict that we will see a rapid acceleration in the use of digital editing systems in just about every facet of our industry.

• The two currently available digital stationary-head recording formats — DASH and Prodigital — will continue to chase market share, and the Sony PCM-3102 (scheduled to be made available to the U.S. market during late January/early February), Twin-DASH 15 ips PCM-3202 (to be shipped in May), and Studer Twin-DASH D-820X (to begin shipment during June) will fuel the move towards digital mastering. In addition, Otari plans to unveil a prototype of its 32-track PD-format DTR-900 at next month's AES Convention in Montreux, while Mitsubishi is scheduled to begin deliveries of the PD-format X-86 two-track in June, and also plans to unveil the PD-format X-400 16-track on half-inch transport at the Montreux AES. Delivery of Sony PCM-3324 and Mitsubishi X-850 machines continue to demonstrate the film-sound, audio-for-video and recording-studio industries' active interest in digital technology.

• Dynamic MIDI control of signal processors will become a strong selling point in the immediate future, and several console manufacturers will look at ways of providing automated control of outboard delay lines, equalizers, reverb systems, compressor-limiters, etc., via MIDI interface, possibly running at multiples of the standard 31.25 Kbaud rate.

• By offering sufficient on-line and hard-disk storage capacity to accommodate upwards of an hour's worth of 32-track/16-bit digital audio, several makers of digital synthesizers will continue to develop useful variants of the "tapeless studio." Once such a capability is made available, watch for digital mixing, EQ and dynamics control to be offered by such up-market synthesizers — possibly beating the traditional console manufacturers to the marketplace with a combination all-digital virtual console and integral multitrack.

On a personal front, I welcome the recent acquisition of *R-e/p* by Intertec Publishing Corporation which, as mentioned in the above statement, will be initiating a monthly schedule beginning with the June issue. I look forward with great anticipation to the creative possibilities afforded by an increased publication frequency.

We plan to retain the current West Coast editorial offices for the immediate future and, in addition, will be helping to coordinate press and media contacts for sister magazines in the Intertec group. Intertec is a 100-year-old company with a consistently successful track record, and currently publishes 14 highly respected business magazines. There is every intention of maintaining *Recording Engineer/Producer*'s leading position as the *Number One Operational Magazine for the Pro-Audio Industry*.

Mel Lambert

Mel Lambert, Editor

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Even though the Pointer Sisters take plenty of chances on stage, they never gamble with their equipment. They use Samson professional wireless systems because of their proven reliability. Samson's long range, no dropout performance gives them the freedom of movement they need in concert. The Pointers have built a career by making smart moves. Like choosing Samson Wireless.

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SANKEN CU-41
MICROPHONE REVIEW
from: **Professor Lowell Cross**
University of Iowa
School of Music

Since I concur with Maseo Konomi [*Letters*, December issue] that the Sanken CU-41 is a very fine microphone, free from most common forms of audible coloration, he and I are in substantial agreement. However, I find some inconsistency between his statement that one can become "entrapped" by relying on a "subjective judgement as to the sound quality microphones are delivering" to one's ears, and his later remark that "the quality of microphones used for CD recordings has become critical, since such quality can be heard." I agree that such quality can be heard, but in listening to CDs, qualitative, subjective judgements are still required. Those of us who actually rely on our ears to make decisions about audio quality will always be making subjective judgements.

I also balk at the idea that "the recent, sudden acceptance of CDs in the market" has ushered in a "rather new way of evaluating microphones." The inclusion of CDs in the signal chain does not reduce the amount of subjectivity required to assess, via the ears, the qualities of microphones; indeed, the use of CDs (or any other consumer-grade storage medium of the present day) can only compound such factors. As reported in the Sennheiser MKH40-P48 review to be found elsewhere in this issue, our judgements have been strengthened when we have compared microphones "live," without having to rely upon even *professional-quality* recording and playback equipment — analog or digital.

Mr. Konomi should have quoted the rest of my comparison of the CU-41 and the Neumann TLM170, since the omitted remarks hardly do the Sanken a disservice: "But, expressed in a different way, the TLM170 could be characterized as having more low-frequency output than the CU-41, with a slightly recessed or withdrawn upper-midrange response, and perhaps offering a bit less extended extreme high-frequency response."

With these relatively minor points aside, I wish to thank Mr. Konomi for his thoughtful comments, and to reaffirm my positive reactions to the CU-41 microphone.

In addtion, certain errors crept into the December 1985 article "Performance Assessments of Studio Microphones," which compared the classic tube microphones to contemporary models. I am grateful to Russell O. Hamm, president of Gotham Audio Corporation, for pointing out that the 6072 dual triode tube used in the AKG C12 and the AKG Tube is not a premium version of the 12AX7/ECC83, but rather a selected *12AY7*. The 6072 is a medium-mu device especially chosen for low "microphonics" (sensitivity to vibration.) The international four-digit number for an industrial-grade 12AX7 *7025*.

An unfortunate typesetting error inadvertantly caused the omission of several lines of text on page 81 of the December 1985 article. The following is the correct text for the first paragraph under the crosshead **"Microphone Evaluations":**

This method of evaluating microphones involves subjective choices. As before, I acknowledge the influence of personal, non-empirical factors in the process, on my part and on the part of our other participants. However, I believe that any problems of subjectivity have been more than offset by the practical, real-world conditions of our recording sessions. We were priviledged to listen "through" these fascinating microphones under realistic circumstances like those encountered daily in the audio industry. The announcement that the German Institute for Broadcasting Technology (IRT, mentioned above) and the German Tonmeister Association (Verband Deutshcer Tonmeister, VDT) are involved in very similar microphone listening tests has contributed to my confidence in our evaluation technique. ∎∎∎

SANKEN CU-41:
ANOTHER OPINION
from: **Michael McLean**
Sound Technician
Burbank, CA

In a letter published in your December 1985 issue, Masao Konomi of Pan Communications, Inc. (export agent for Sanken Microphones) suggests that the subjective comparison of the Sanken CU-41 to the Neumann TML170 is misleading and/or invalid, on the basis that Sanken engineers restrict themselves to an absolute minimum coloration design philosophy.

What Mr. Konomi suggests is completely invalid for the simple reason that *all* microphones with a headgrill structure have, *inherently*, a *very substantial* amount of upper frequency coloration caused by reflections off the grill structure.

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Each MC input channel has a 20 dB pad and gain control with peak LED, three-band EQ with sweepable mid-range, two pre-EQ and pre-fader fold-back sends, two post-EQ and post-fader echo sends, pan control, group 1-4

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tronically balanced with XLR-type connectors. And there are insert patch points on all input channels as well as on the groups.

Talkback facilities include a headphone jack, cue/phones level control, talkback assignment switches, and a mic input XLR connector with an input level control and switch.

Yet with all these features and flexibility, the MC Series mixing consoles are compact and lightweight. As well as affordable.

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*U.S.A. suggested retail price. Canadian suggested retail prices are $4,995 CDM for the MC2404, $3,695 CDM for the MC1604, and $2,995 CDM for the MC1204.

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The Sanken CU-41 (and the TML170) have a predominant head grill, and it is therefore ludicrous to suggest that it is a minimum coloration design.

The name of the game with these microphones is "pleasing coloration," rather than "minimum coloration."

In fact, *true* low-coloration designs have been available for years, but have only attracted a limited segment of the market. One example of such a design is the Neumann KM-83 and KM-84 types. Here, a great effort is made in finding a good balance between minimum head grill effect, and adequate protection of the capsule from environmental damage.

The most extreme example of ultra-low coloration design would be a B&K quarter-inch measurement microphone with the protecting grid removed. This device has virtually zero coloration in the audible range, but is very easy to damage due to the exposed diaphragm.

Anyone who has done repair work on studio microphones is well aware of the profound change in coloration that takes place when the headgrill is removed.

For those who wish to prove this for themselves, I would suggest the following experiment: select a matched pair of, say, Neumann U87s, and a KM-84. Use the 87s in cardioid mode to match the KM-84, and set them up side by side. With both head grills in place, the 87s will sound alike, and quite different on the high-end from the 84s. If the head grill of one of the 87s is carefully removed (do not disassemble a U87 that is still under warranty, because of the wax seal), it will be easy to hear that the modified U87 sounds very different from the unmodified U87, and indeed, sounds much more like the KM-84! You can rest assured that the same sort of results would be observed if the 87s were replaced with CU-41s. *All* head grills have coloration.

I feel that a very important element in the success of a microphone maker, that offers a "high-end" professional product, is an image of impeccable scientific and engineering integrity. Over the last 30 years, I have observed that the leading such maker, Neumann, has always maintained the highest standards in keeping illogical and unscientific doubletalk out of their advertisement and product literature.

I feel that Mr. Konomi's letter puts Sanken, a company that probably has a very fine product, in a very bad light in this regard.

Subjective testing, such as that by Professor Cross, is the *only* way to try to measure "which coloration is most pleasing" other than, of course, the marketplace, where headgrill designs tromp the low-coloration designs year after year. In view of this, Mr. Konomi's suggestion that Professor Cross is "entrapped" in his good work is laughable and tragic. ■■■

## VIRTUAL CONSOLE TOPOGRAPHY
from: **Dan Tinen**
Wheeling, IL

Your article on virtual consoles (October 1985 issue) was excellent. It has spurred me to offer some comments of my own:

An analogy with the keyboard world: Synthesizers started out with manual patch cords, (ARP 2600), advanced to switched routing (Minimoog), then to computer memory of dozens of knob settings (Sequential Circuits Prophet 5), and then reduced all the knobs to a single virtual control (Yamaha DX-7, Roland JX-3P, etc.). Even though the keyboard players missed their knobs at first, those who knew how to program got used to the focused thought process of choosing parameters before twiddling with them. People who knew how to get what they wanted out of a synthesizer accepted the new designs; those who had gotten their favorite patches by accident (when manipulating controls they weren't thinking about) disliked them. Virtual consoles will probably cause a similar disruption among engineers.

The good news: Old instincts get replaced by new ones through experience. Keyboard players now routinely alter programs and replace functions on DX-7s with a single fader, a "yes-no" switch, and a small alphanumeric display. "Where are all the knobs?" is no longer the main complaint. Similarly, virtual consoles will probably meet with

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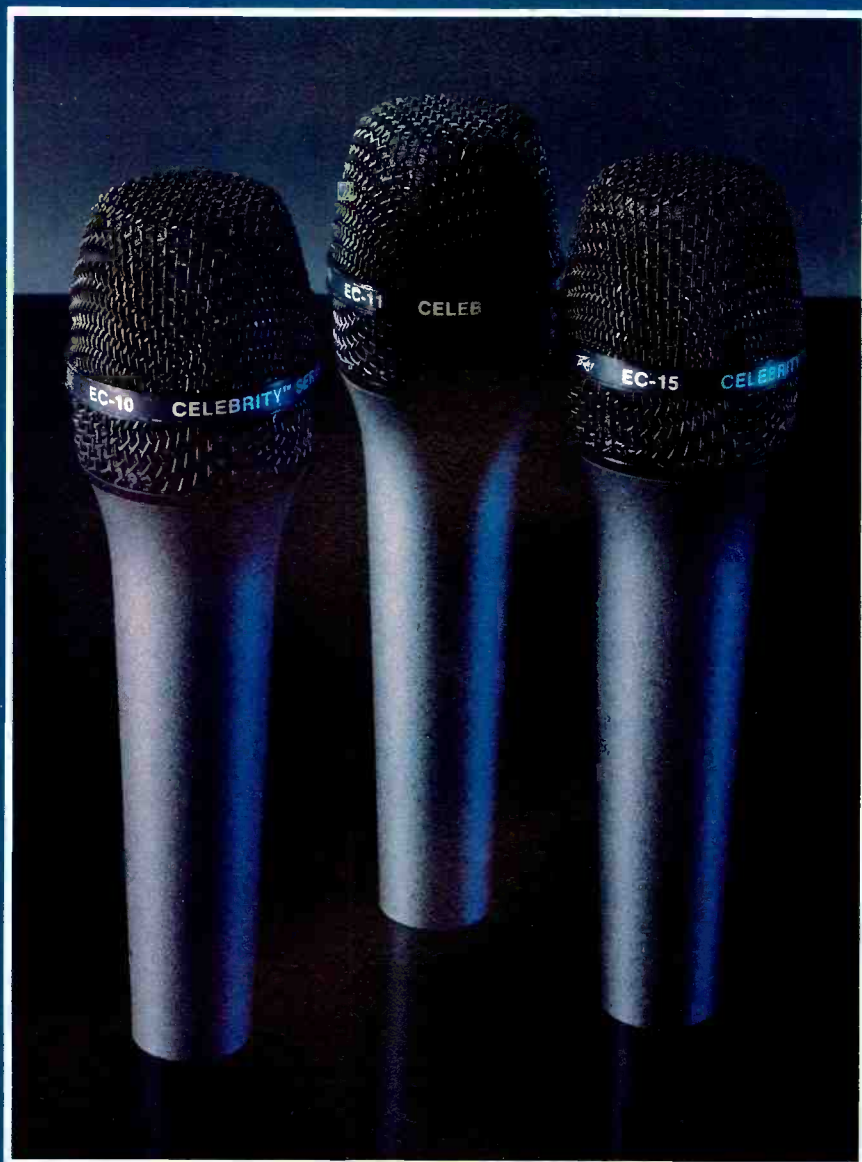
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LETTERS

resistance until their advantages become apparent . . . and the users get past the initial learning curve.

On using the control surface to display settings: Removing the immediate feedback of settings on current consoles may not be a totally bad thing. As opposed to moving controls according to some prejudice ("I don't like the 3 dB boost on the 15 kHz of the high-hat"), dealing directly with the *audible* consequences of moving a control is what our business is really about. How many artists, producers and engineers "listen with their eyes" — i.e., base their opinion of a mix on looking at the board instead of listening to the monitors? Having a virtual console where controls can be "hidden" could actually be an advantage.

On standardization of formats: Standardizations of data formats between consoles would be nice but, given the wide variety of consoles out there, it could be sort of like programming a DX-7 through the controls of a Prophet 5. More exciting and useful, perhaps, is standardization of control room acoustics by eliminating the console and making it more like a typical listening environment. (Personally, I don't listen to music in my living room by facing my speakers with my elbows on a large table!) Sometime in the future, I look forward to doing final mixes in an IEC-standard listening room, sitting on a couch, using a mouse, light pen, or small lap keyboard to manipulate images of controls projected on the wall between the speakers. The EditDroid (why wasn't it included in your article?) looks like the first of this kind of console.

I'm looking forward to virtual consoles, though a "snapshot" of most control settings will be enough for me (as opposed to frame-accurate dynamic updates). My work with acoustic musicians doesn't require sweeping EQ or pan in the middle of a mix, but clients often want it "just like last week's mix, but with more high end in the bass."

■■■

TWO-BOX LOUDSPEAKER SYSTEMS: A CORRECTION

from: **Dave J. Beecham, president**
Audio Techniques, Inc.
Calabasas, CA

Regarding David Scheirman's article entitled "Design Evolution of Two-Box Packaged Loudspeaker Systems," published in the December 1985 issue, please note that there is an error in the quoted crossover frequencies of our system:

• On page 59, under "Systems Electronics," the crossover frequencies should read 150 Hz, 1 kHz and 7 kHz, and not as shown.

• On page 61, under "Table 1: Comparison of Transducers," the crossover frequency under Low should read 27 to 150 Hz and, under Mid, 150 Hz to 1 kHz.

My compliments to *R-e/p* and to David Scheirman for an extremely well written article.

■■■

SONY APR-5002 REVIEW

from: **Stanley P. Lipshitz and**
John Vanderkooy
Audio Research Group
University of Waterloo
Ontario, Canada

Not just the Sony engineers, but we too were rather perplexed by Peter Butt's frequency- and phase-response measurements on the Sony APR-5002 analog tape recorder (*R-e/p*, August 1985 issue and *Letter* in the October 1985 issue). We believe that the measurement procedure being used requires more thought, for it seems to us that it may not be accurately measuring quite the parameters which the author-intends. More specifically, we would like to comment as follows:

• One trouble with squarewave excitation is that it produces a rather sparse input spectrum. In fact, with the squarewave frequencies and DFT sampling rates used by Mr. Butt (which appear to be frequency locked to produce precisely one squarewave cycle in the analysis window), every second frequency bin in the DFT analysis would have zero excitation were it not for the window presumably being applied to the data. For example, a Hann window spreads this power in a uniform manner between bins.

Another problem is that, although the squarewave spectrum falls at 6 dB per octave, the sharp (non-band-limited?) edges can produce high-frequency overload or the effect of overbiasing as pointed out by Mr. Butt. The spectral trend of a squarewave may approximate that of music, but its temporal characteristics do not. We wonder whether this might have been a problem, especially at the 7.5 ips tape speed, where a rather drastic high-frequency rolloff was measured. We agree with Mr. Butt that a broader-band signal than a slowly-swept sinewave is desirable for more meaningful tape-recorder transfer function measurements.

We tend to favor the low-level broadband noise as the excitation. This avoids the objections listed above. A flatter input spectrum can be produced if a pseudo-random (binary or Gaussian) noise (PRN) source clocked at a rate preferably locked to the FFT sampling clock is used as the excitation signal.

If a single-channel rather than a dual-channel FFT analyzer is used, it must also be possible to trigger the PRN source to commence at the same point in its sequence at each measurement. Many FFT analyzers offer such a noise output for this purpose. The overbiasing and overload problems can be avoided by bandlimiting the noise before feed-

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ing it to the tape recorder, and keeping its level well below tape saturation.

• Mr. Butt appears to be using a single-channel FFT-analyzer for the measurements, and computing the complex quotient of the output and input spectra of the system under test. This is quite reasonable on normal systems, but needs to be used with care on systems which display time jitter, such as even the very best analog tape recorders. For, if any signal averaging is attempted (and it is not clear whether this is being done), the transfer function magnitude thus computed tends to "average down" at high frequencies due to the jitter-produced phase fluctuation between individual measurement records. This is particularly evident at the lower tape speeds (e.g., 7.5 ips) where very considerable errors will be obtained above a few kiloHertz due to the motional irregularity.

On a dual-channel FFT analyzer, the drastic reduction in the measured coherence function at high frequencies will warn one of this potential error but, with a single-channel measurement, one may not be aware of the difficulty.

Indeed, correctly measuring both the magnitude and phase of the transfer function of an analog tape recorder is extremely difficult.[1] The 7.5 ips measurement of Figure 7 [August 1985 issue, page 136] especially seems to show severe high-frequency rolloff. Is this due to jitter combined with averaging? Similar doubts apply as to the correctness of Figures 8, 9, and 12 thru 14 — not just the magnitude but also the averaged phase response is affected by the jitter.

A correct frequency-response magnitude can be obtained if one computes rather the *transmissibility*: that is, the (real) ratio of the output and input averaged power spectra. Most dual-channnel analyzers offer this function.

Obtaining a meaningful phase measurement is harder. One way round the jitter problem is to use a very brief excitation signal, such as a low-level impulse, and then do signal averaging. In this way one can obtain good coherence to beyond 20 kHz, but it is still difficult to achieve; tape weave, for example, degrades the results significantly.

• The phase curves shown in Figures 7 thru 9, and 12 thru 14 are not correct. The progressive phase lead at high frequencies represents an *acausal* measurement of the non-minimum-phase analog tape-recorder system. As Figure 11 shows, the whole measured high-frequency group delay is negative. This is a sure sign of an incorrect (acausal) input/output time delay correction during the measurement. In fact, the constancy of the high-frequency group delay shows that the phase correction performed by the all-pass equalizers is accurate.

By the way, an all-pass equalizer produces pure phase *lag* and cannot cause the phase lead shown in Figures 7 thru 9, and 12 thru 14. It *is* correct to state that the analog record/reproduce losses

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are phaseless[1]. The all-pass equalizer is needed to correct for the phase lead which accompanies the high-frequency boosts of the record and play electronics. These boosts in turn are needed to correct for the considerable high-frequency losses inherent in the process.

The squarewave traces in Figure 10 show this phase compensation to be accurate. An indicator of proper high-frequency phase compensation is the equality of the pre- and post-transition overshoots on the waveform.

To summarize: the input signal used appears not to be optimally chosen; the averaging — if used — is a cause of high-frequency measurement error; and the phase responses are wrongly time-delay corrected. We would prefer to see alternative methods used for tape-recorder measurement. In particualr, broadband noise would be a better signal than squarewave, and if the phase response is required, the (low-level) impulse response averaged and transformed by a single-channel FFT analyzer can be used. We hope that these comments prove useful. ∎∎∎

References:

1. J. Vanderkooy and S. P. Lipshitz, *Polarity and Phase Standards for Analog Tape Recorders*; presented at the 69th Convention of the Audio Engineering Society, Los Angeles, May 1985 (preprint #1795).

Editorial Note: Due to pressure of work, Peter Butt was unable to prepare a suitable reply to the above letter. We plan to publish his reply in the April issue of *R-e/p — ML.*

EXPOSING AUDIO MYTHOLOGY

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

by John H. Roberts

This month's column will spotlight some of the current considerations for studios that are contemplating what many would consider to be the inevitable transition from analog to digital consoles.

Digital Consoles for Digital Recordings?

Some people have suggested that "analog" consoles are somehow not up to the task of making "digital" recordings. While there are any number of less than crystalline sounding analog consoles out there corrupting digital, and for that matter, analog recordings, they are *not* typical. There will probably be a few nasty-sounding digital consoles out there as soon as we get enough people making them. For the sake of this discussion, let's look at the typical professional console (in good working condition).

As I have discussed a few times in prior writings,[1,2] the difficult tasks to perform in a recording console are the amplification of microphone-level signals, and to some lesser extent, the summation of several of these amplified and processed signals into a two- (or four-) channel final product.

The digital console offers a theoretically perfect summation but, as I've discussed in reference #1, analog consoles can approach these theoretical limits, and easily exceed the dynamic range of present digital systems. For smaller (less than 10-input) mixers, even simple summing circuits typically will not be a limitation.

What this leaves us with is the microphone pre-amplifier as the remaining performance limitation. With the possible exception of "the boys (and girls, of course) at Bell" I don't expect anyone is seriously working on a digital microphone, and probably they are struggling just to get "voice-grade" performance.

To directly accept the low-level output from typically professional microphones, an A/D convertor would need something on the order of 25 to 30 bits. Although 16 bit PCM digital is not a technological brick wall, the current state-of-the-art is only two or three bits more, making direct microphone level input out of the question (for now).

Thus, the performance of both digital and analog recording consoles will be limited by an analog microphone pre-amplifier stage. As is often the case, the absolute performance realized will be a function of how well this circuit block is executed, rather than some intrinsic technology.

Curiously, digital recording consoles are much more likely to benefit heavily over-produced multitrack recording, than the minimalist, three or four microphone, purist recordings. Keeping the signal within the digital domain during the numerous bounces, and even performing some of the effects processing on the bit-stream itself, will result in a

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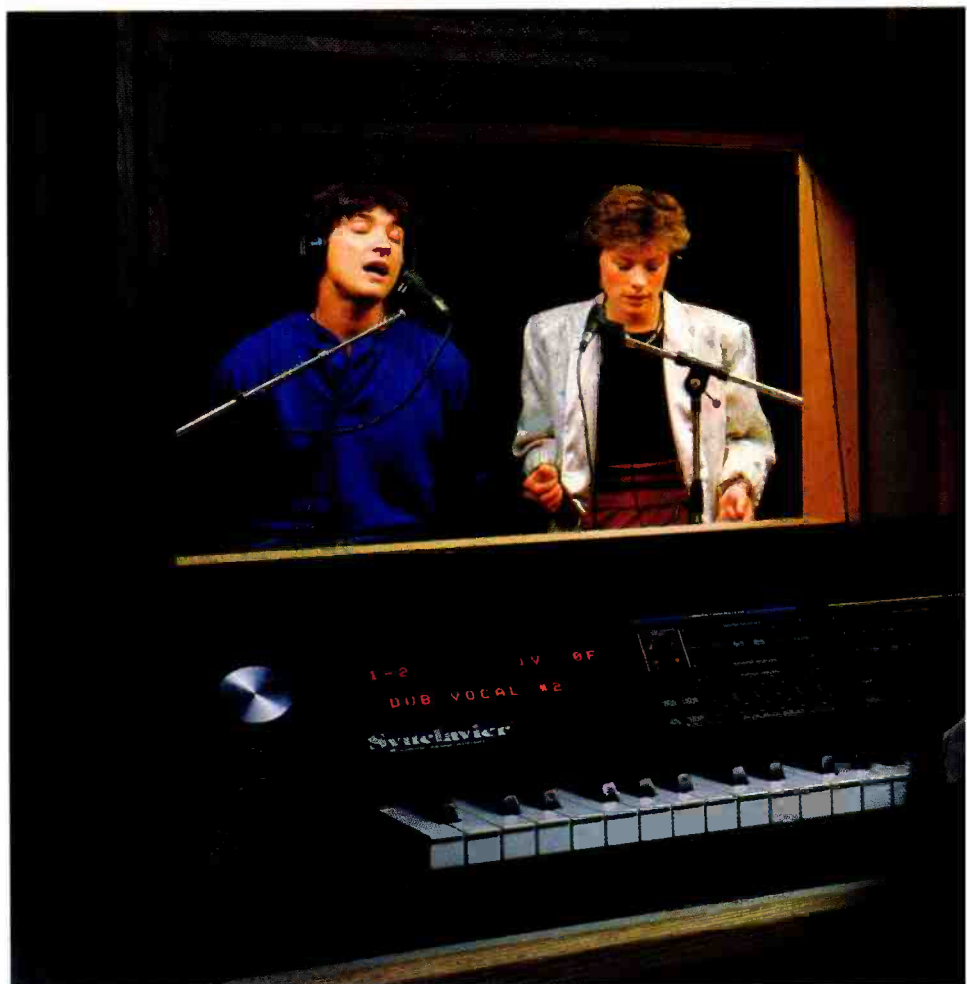
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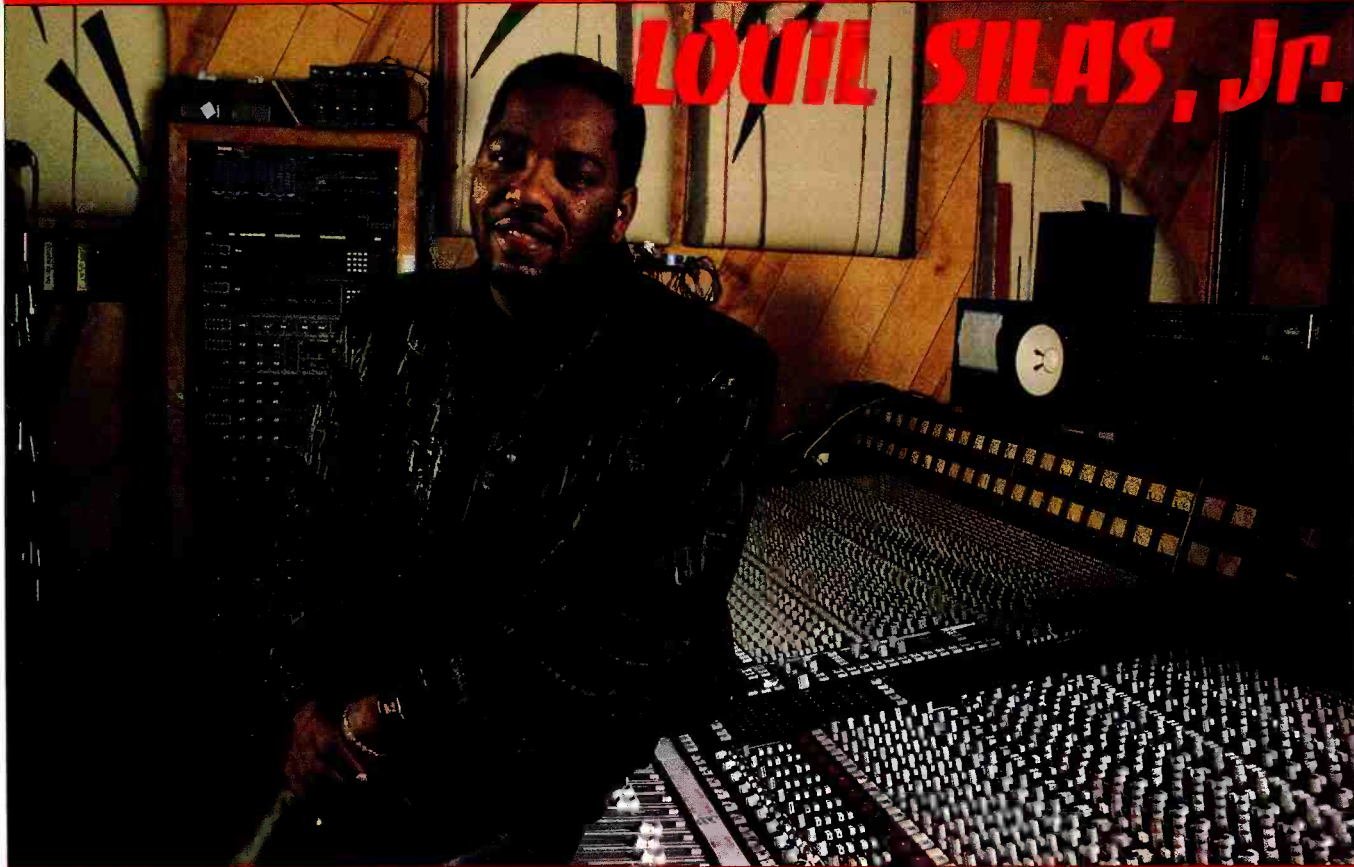
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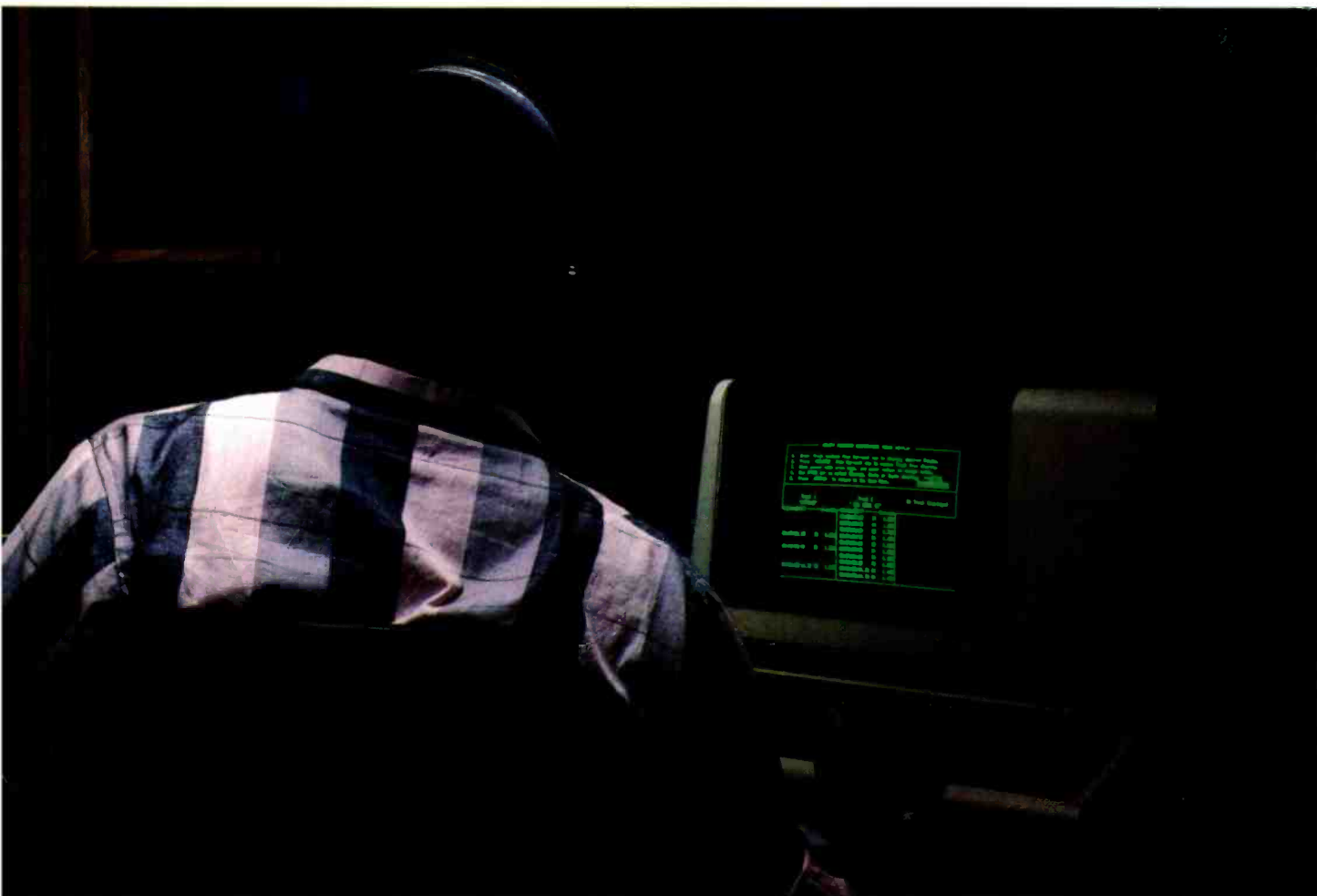
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Synclavier operator captures continuous live vocal overdub

All photography by Elizabeth Annas

LOUIL SILAS, Jr.

SPECIALIZING IN DANCE-SINGLE REMIXES
Interviewed by Ralph Jones

As A&R director for MCA's Black Division, Louil Silas, Jr. has risen quickly to become one of the most prominent single-remix producers in our industry, while simultaneously acting as a prime force in affecting the label's recent renaissance. Like many of his peers in the field of remixing dance singles, Silas started as a disk jockey at a Los Angeles-area dance club. In subsequent promotional positions — first for A&M Records and then for MCA — he gained invaluable first-hand experience in the marketing of contemporary music. In 1984, given the chance to remix the single version of Alicia Myer's "You Get The Best From Me," he entered Larrabee Sound Studio with session engineer Taavi Moté, and walked out with a Top Five Record.

In the two years since their first single was released, the Silas/Moté team has mixed a series of hit singles, including New Edition's "Cool It Now"; "Oh Sheila," and "Digital Display" from Ready For The World's self-titled Gold album; Patti LaBelle's "New Attitude" and Harold Faltermeyer's "Axel F." from the *Beverly Hills Cop* soundtrack; plus the current Klymaxx ballad, "I Miss You."

When *R-e/p* met with Louil Silas, the extremely busy young executive was hard at work on a single version of "Crush On You" by The Jets. The following in-depth discussion ranges over a variety of topics, from the remixing process to current A&R developments at MCA Records.

R-e/p (Ralph Jones): Let's begin with a few details regarding your background. How did you reach your present position with MCA?

Louil Silas: I started in television: my degree is from Cal State Dominguez Hills, with a major in teleproduction. While in school, I worked as a page at KTTV [a major Los Angeles TV station], and then got a gig as a courier. At the same time, I worked as a DJ at house parties, so the music thing was in the back of my mind. From courier, however, I went on to be a staff writer on a PBS television show called *The Righteous Apples*. I wrote for that show for a year, and then it was cancelled. I was looking for something to do when a club opened up — a new disco in the Culver City [Southern California] area, called Contempo's. I started DJ'ing there, and ended up staying for three years. In that time, I established a reputation as one of the "baddest" guys in the city: It was one of the first clubs that record promoters would hit to get a gauge on what was hot.

R-e/p (Ralph Jones): While working

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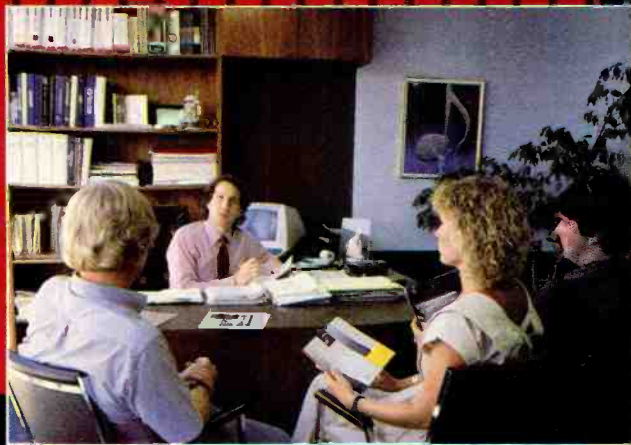
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LOUIL SILAS, Jr.

as a club DJ, were you doing Scratch?
Louil Silas: Back then, no. I was just concentrating on mixing from one song to another — that was the big thing then. If you could do five hours and take the audience on a continuous high, you were good. Around 1983, when I stopped DJ'ing regularly, Scratch was really starting to happen, and I got in on the fringe of it. But I didn't like it ruining my records, so I wasn't that high on it! [Laughs at the memory.]

Anyway, I was interested in getting into the industry. I met a lot of record promoters at the club and, finally, one of them told me that there was an opening in promotion at Atlantic Records. I applied, got the job, and worked there for a year as a local promotion representative. Then, I moved to MCA in '83 as West Coast regional promotion manager.

At this stage, I was helped enormously by a man named Clarence Avant — the president of Tabu Records, and sort of a guru to black record executives. Clarence is one guy who's looking out for young black executives' interests, because he was in the industry before it was what it is now, and he wants to be sure that everybody gets their just rewards. He personally negotiated my contract with MCA, which is the reason it took nine months rather than nine minutes!

Shortly thereafter, Jheryl Busby came to MCA. Now, during my time as a DJ, when Jheryl was at A&M, I used to bug him to let me do remixes. When he came to MCA, knowing that we had no A&R input from the West Coast, I let him know that I was available to do remixes, and that I thought I could do them *well*. He finally gave me the opportunity, in April of '84, to come into the studio. I met [session engineer] Taavi Moté through Jimmy Jam and Terry Lewis, and he and I did my first remix, which was a song by Alicia Myers called "You Get The Best From Me." That first remix went Top Five on the Black charts.

My title now at MCA is Director of A&R, Black Music Division, but my job entails a *lot* more than just choosing singles. I like to get involved with the music and, hopefully, make it better! I think that, from my club experience, I have a feel for what's going

on, and I go out enough and talk to enough people to get a gauge on what's hot.

R-e/p (Ralph Jones): And you now do all of the single remixes for MCA?
Louil Silas: I do most of them, because when I've hired other people to remix, I second-guess them. Then, I have to live with any disappointment I may feel about the result, and I have to talk to managers and artists who want to know my opinion of the mix. Usually, they want to know why I didn't do it! I've kind of spoiled people on the label, in that they know that I've had a degree of success, and they want to be a part of that.

R-e/p: Let's get back to the fundamentals: What, to your mind, is the
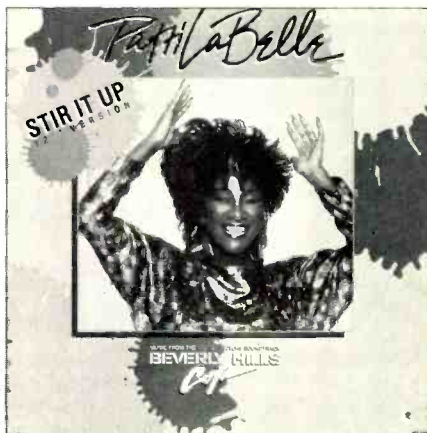
intended use for a single's remix? Is it produced primarily for radio airplay, or dance clubs, for example?
LS: I do several mixes, actually, in different categories. One is an extended remix, which is designed solely for clubs: if you heard it on the radio it might be monotonous, because there are some things that are obviously stretched out. When I do those mixes, I'm thinking *strictly* of the "pulse" of the club, and what the DJ would want to use.

My second mix is a radio edit of the extended club mix. In that version, I'm giving radio what they want — a shorter intro, shorter break, and so on — but all the elements of the club mix are there, so that the people who heard it at the club will think that this version is just as attractive.

The third version is the single edit, which is a tool that radio uses in AM or PM drive-times, when they don't have a lot of time to play longer versions. Kids buy a lot of singles, so I try to use the elements of both of the other

versions, to make it attractive for them to buy.

Then, I do "play" things, like dub mixes or "a cappella" versions, which are designed for DJs to play with bonus beats to segue from one part of my mix to another. When you go into a club with one of these play mixes, the DJ will play the record, for something like 20 minutes. The exposure is great if you give them a lot to play with! When I was a DJ, if I saw five or six mixes on a disk, and the record was hot, the song got a *lot* of play. So, on the 12-inch release, with all those versions, I'm thinking mainly of club exposure. It's a proven fact that we can break records through clubs, so the more exposure I get, the better our sales are!

R-e/p: How many versions of a particular song do you usually release on a 12-inch version?
LS: The most is five. As I explained, in addition to the extended mix and radio edit, I make different fractions of those mixes — like a cappella, or what have you. Again, it's all designed for exposure, but when the public is able to purchase the 12-inches, I've found that they also like them. The single edit also goes out on a seven-inch.

R-e/p: What about the radio edit? Is that released separately?
LS: That's included on the 12-inch. The way I make my 12-inches, the radio edit opens up the B-side. When the people at the radio station get the record, they say, "Well, man, the A-side is nine minutes! I can't play this!" Then they flip it over, and they say, "Louil's done it again! There's my version!" They should recognize that now: it's been happening at MCA for a year; that's my format.

Some stations play the long versions of everything I've done. My longest mix was nearly 10 minutes; it was a song by Network, called "Out Of Danger," and it's out currently. I was surprised the stations even carted that one up! I like long mixes, though: I'm trying to reach Prince's record. I think the "America" mix is over 21 minutes long — I'd like to do a mix like that!

R-e/p: Would it be safe to say that, in general, you simply extend the original single for the 12-inch versions?
LS: It depends. More often than not, I do. Usually, when I hear what the

LOUIL SILAS, Jr.

artist and producer think is the finished version, I'll hear certain elements underneath that I think should be accentuated. To get to the point where those elements are brought out usually makes it longer: the A-sides on my releases are normally about seven minutes. The song we're working on tonight ["Crush On You" by the Jets] is only five and a half minutes on the album, but it'll probably be about eight minutes long when I'm finished with my extended version.

R-e/p: How do you go about extending a song by as much as 50%?
LS: I find the pieces that I want to accentuate, and mix them to two-track individually, then I have Taavi [Moté] splice them together. Your engineer, if he's good, can make it sound like that's the way it's supposed to be.

You bring things up, and mute other things to make it sound different: like, if I want to do a bass breakdown, I might have to take out some percussion to accentuate it. I like that part of it. In R&B music, especially, the breakdown is the part that makes the people just go crazy! As a DJ, I *loved* breakdowns.

R-e/p: Do you also use digital sampling to capture elements and move them around in the song?
LS: Oh, man, *all* the time! In fact, my nickname at one point was Louil "AMS" Silas! When I first came here to Larrabee[Studios, Hollywood],and Taavi showed me the things that different pieces of outboard equipment could do, I was amazed. I love the AMS [DMX 15-80S digital delay/sampler] and the [E-mu Systems] Emulator.

Usually, I use sampling for vocal phrases that I may want in a different part of the song. Or, for example, the song we're working on tonight — "Crush On You," by the Jets — has a tom fill at the end that I want [to place] in the first verse. So, we'll be using the AMS to move it.

As I learn more about what each piece of equipment does, it's expanding my capabilities. I used to just come in and mix from the recorded tracks; now, I'm bringing in extra musicians. I have a guy that I work with by the name of David Irwin, who has just about every piece of keyboard equipment ever made. I can call him in, and he'll bring his Emulator, [Yamaha] DX-7, Moog or whatever, and give me what I want. So, as I'm growing into this thing — wanting more sounds; wanting to experiment more with the mixes — I'm just using

— Louil Silas, Jr. and Taavi Moté—

the equipment more.

Hopefully, it's sounding a bit more advanced, but I'm not really into trying to impress people in the industry with what I'm doing. The people don't care about sampling or what have you: all they want to know is that it sounds *good*. So, I'm just trying to make it sound good, and make it attractive for people to buy.

R-e/p: Do you often find yourself replacing sounds on the original tracks — making a new kick sound, for example — rather than just moving elements or adding parts?
LS: Yes, we've done that. It seems like I did more of that kind of thing on the earlier remixes, but I've brought people in to replace guitar parts, and we have a sample tape with keyboard sounds, drum sounds, and what have you. For example, I sampled from the "Oh Sheila" track. That song went number one in the nation, so I figure that if I want a good kick drum sound, what better one to use than the "Oh Sheila" kick? That sound has been used on a couple of remixes now!

R-e/p: You've talked mostly about the "feel" of the track. Do you also involve yourself with technical aspects of the remix?
LS: Right now, I concentrate on feel, because I'm going from the club experience: that's what I draw from for my mixes. The audience doesn't *care* about technology — they care how it *feels*. So, when I'm doing these things, I become the DJ and the audience. If it excites me, it should excite them, too.

I also have a lot of input from people at MCA, and from friends that just

love music. Usually, before we go to a final mix, I get a lot of opinions. Of course, it depends on the time and the budget I have to work with: a lot of these remixes are charged against the artist. Since I'm spending their money, I try to be real frugal. But we have it down to a kind of formula now, so I should be able to do a remix in about 20 hours. In that amount of time, I should have everything I need for all the versions that I want.

One exception to that was "Digital Display," by Ready For The World: I did that remix over a period of about two weeks. The group was in town, and we brought them in to do overdubs. Then I brought in [keyboard player] David [Irwin] to do some things, and that thing ended up costing about $18,000. Real high!

R-e/p: Are those additional studio costs for the remix charged against artist royalties?
LS: Usually. It depends on the artist's contract, and how good he is at negotiating. Nine times out of 10, the record company, rather than the artist, calls for the remix — at least in my experience with MCA. But the cost of the remixes I'm doing now is between $6,000 and $8,000. If it's based solely on royalties, the artist will make that back, because the remixes have been selling at a pretty good clip!

R-e/p: If you concentrate mainly on the "feel" of the cut, I assume that you rely on your session engineer, Taavi Moté to handle the more technical aspects of the mix.
LS: That's right. Taavi gets the primary sound together, then my thing is the *balance* of those sounds. Of course, once Taavi has it the way he wants it to sound, it's not always the way I want it. So, we make adjustments when I come in. But I have so much other stuff to do during the day that I leave the track with Taavi for about three hours to let him get the basic sound together, and then we move from there.

Taavi is *good*. You know, a lot of guys aren't using engineers that are sticklers for detail — but, to me, the engineer is *the* most important part of any project. If you have somebody who's willing to go to the "nth" degree to make your song sound better, you're going to win! I'm lucky enough to have that kind of a guy working with me, and our mixes will *always* sound good, whether the record's a dog or not.

R-e/p: You said earlier that the record company generally calls for the remix. When do they decide to release a

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> **"A remix should never sound worse than the LP cut: it should always sound *better* — I don't care who does it, or who the engineer is. It should always hit the listener a bit differently, like a revival."**

LOUIL SILAS, Jr.

remix? What are the indicators?
LS: First, we have a promotion-staff conference call. Then the promotion director, Ernie Singleton, will come back to me with feedback from the field staff as to what the next single will be. I'll then talk with my boss, and between all those opinions should come a decision on the next single. It's not solely my decision. Once we make the decision, however, it's mine to live up to.

R-e/p: But why go to the extra trouble — and cost — of remixing the track? Why not just release the album cut as a single?
LS: Well, nine times out of 10 — if it's an up-tempo single, and the LP is popular — it's probably been played off the LP as an album cut. So, we know that the life of that song might not be so great once we release it as a single: once a song's been played as an album cut, it's been burned, so we try to infuse some "fresh air" into the situation with a remix.

Sometimes, we'll do an edit of the LP version, and release that as a single to radio, and then release the remix later. Of course, on a second single, I *never* want to come out with the remix right out of the box. I always want the remix to be the next level — to boost the tune, and prolong its life. We have that situation with "Digital Display," by Ready For The World: that cut was played heavily off of the LP. We released the album version as a single, and it was the most-added single. It came on the charts at 56 and has gone to 40; it'll go to the high-20s before we release the remix. Then, that breath of fresh air might take us all the way to Top Five.

R-e/p: Do you think that there's something unique about the remix that would help to move the song higher on the charts? Not simply that it's different, but something specific about the feel and the sound?
LS: Well, the feel is always different, even in the pressing itself. You're pressing a 12-inch, so the grooves are bigger. The lows are lower, and the highs are higher. Plus, you're always trying to improve that sound. Even if you think you got the maximum sound on the album cut, there's always something *more* you can do.

A remix should never sound worse than the LP cut: it should always sound *better* — I don't care who does it, or who the engineer is. It should always hit the listener a bit differently, like a revival. Even if the record is in recurrent at most radio stations, if you have a hot, innovative remix, you'll see the record added to the play-list again.

R-e/p: What then makes for a hot remix?
LS: Whew! [Pauses and collects his thoughts] Concentration on what the people want to hear. You're able to draw out something that, on the LP cut, maybe you had to condense because you only have 21 minutes per side. There might be part of the record that, when you hear it on the LP, makes you say, "Golly, I wish I could make that longer!" On the 12-inch, you can do that, and it makes the song more interesting. I know that some people may find long versions monotonous, but to me — and, I think, to a majority of the listeners — attention to those details makes for a better record.

R-e/p: Do you also do similar things to the radio-edit version? Would you draw out elements in the same way?
LS: I draw out as many as I can — between five minutes and 5:30. Again, I was in promotion before I got into A&R, and I know what radio wants. Sometimes, I talk with radio people and get their opinions. When I was remixing Patti LaBelle's "New Attitude," we started off a cappella: [sings] "I'm feeling good from my head to my shoes." I asked a [radio staion] programmer how she would program that record, as opposed starting it off with rhythm, and then the vocal. It's just better for that a cappella intro to be played at certain points in their programming day. I wanted to know whether I was going to be hurt by starting off the record like this, or if it was going to help me. With that record, it was a *definite* help, because it came out of nowhere, and got a lot of attention. The other song would be fading out and, all of a sudden, Patti would come on.

R-e/p: The Patti LaBelle remix must have sounded kind of startling, being backward from the way most tunes begin?
LS: Right. Of course, I didn't want to blow it, because it was an important record! When I go into the studio to do a remix, I say a prayer every time, asking *not* to blow it! You can blow a record as easily as you can make it better, just by pressing the wrong button.

You know, a lot of times people think they're being creative, but they may just be destroying the feel. Luckily, there's never been a case where I've just completely blown it because of some concept that I had, but I have had big records that I thought I could have done a better job with. I won't name the song, but I had a Top Five record that I hated from the moment it was released: I wanted to redo it. But it just caught on and, to this day, I can't believe that the record did what it did. I heard some things in there that I *know* were caused by fatigue, because it was a time that I was traveling back and forth to New York, and I went right from the studio to the airport. But that happens very rarely.

R-e/p: What, for you, is most important element in the remix?
LS: I have basic criteria for things that I want to do. First of all, I want to start the remix off with the most exciting intro that I can possibly do, within the confines of the particular song. I want people to put the needle down and, when the thing comes on, say, "Whoa! I like that!"

After the intro gets their attention, I build up the body of the song, and then I want something in the middle that makes them go crazy. Usually, that's accentuating the groove; sometimes, it's repeating a particular phrase that the artist sings. There's such a variety of things that you can do.

I would say that for most songs, people don't even know 80% of the lyrics. But they'll know the groove and, if you accentuate that groove, nine times out of 10 it'll be a successful record. So, I would have to say that the *groove* is one of the most important aspects. That, plus the intro and the breakdown. Those are the three most important parts. *continued overleaf —*

CONTRASTING CONVENTIONAL MIXDOWN TECHNIQUES WITH DANCE REMIXES

A Conversation With Louil Silas' Regular Session Engineer Taavi Moté

Engineer Taavi Moté, who currently serves as Louil Silas' right-hand man and collaborator in the studio, began his career as a guitarist in a succession of high school bands. Upon graduation, the transition to studio musician introduced Moté to the environment which, for him, proved to have an irresistible appeal: "All I had to do was get in the studio once," he recalls, "and I fell in love with it. The atmosphere was so conducive to creativity!"

The Recording School Of America, and a stint as an assistant Larrabee Sound Studios, Hollywood, provided Moté with the education in studio techniques that has enabled him to become a much sought-after independent engineer. He is a dedicated "Solid State Logicman" and, in addition to Larrabee, also works regularly in such Los Angeles facilities as Encore Studios, Record Plant and Solar Studios. His recent credits include self-titled albums by Ready For The World and New Edition; Patti Labelle's contribution to the *Beverly Hills Cop* soundtrack album, "New Attitude" (nominated for two Grammy awards); and the Klymaxx album, *Meeting In The Ladies' Room*, which features the current Top-Five single, "I Miss You."

We began our discussion by asking Taavi Moté to contrast the characteristics of a dance-single remix with those of an album mix. "One of the biggest advantages of the 12-inch single," he replies, "is that you can mix more low-end content. On an LP, an equivalent amount of low-end is going to take up too much groove space, so that, on an album which is 20 or 22 minutes long, the grooves will start getting too close together. You can also spend more time with a rhythmic break than you would on a single, play around with the melody longer, and expand a musical thought. That allows you to make the record more *exciting*, by bringing out things that may have been lost in the LP version.

"I find that with a remix, being 'cold' to the song — not really knowing it that well — is an advantage: you bring out things that somebody else might have pushed back, either out of habit or preference. You can get into the 'feel' all on its own, whereas you couldn't do that on an LP cut. Also, you can have much more movement — more explosive things in the mix. That's partly a matter of panning, but it also involves making things come back and forth from front to rear; things like that can really make the music explosive."

We surmised that outboard equipment plays a significant role in the process — an impression that Taavi confirmed. "Outboard equipment is one of the most important things for any remix: it allows you to enhance what's there, change things around, and even make a track sound totally different from what it sounded like originally. In digital delay lines, I prefer the AMS [DMX 15-80S], because I can feed a signal in, and have the same quality of sound come out. Their RMX-16 reverb is wonderful: the reverse programs and the non-linear program, in particular, have been quite popular in the past couple of years. And the ambience programs are great for taking a synthesizer track or drum computer, which might have a one-dimensional sound, and turning it into a three-dimensional-sounding instrument.

"We also do a lot of sampling. For example, sometimes Louil will want to hear something moved from the top of the song to the last verse, or we'll have to take a chorus part and move it into the whole vamp over and over again. In the old days, we would sample onto two-track and try to roll it back in, but now we have the luxury of using sampling equipment, especially the AMS 15-80. We dial something into it, edit it down, and then either trigger it manually or use something rhythmically from the multitrack tape. It's definitely made the mixes *much* easier, allowing us to be more creative.

"I also enjoy using the BBE Model 202R, a sound processor made by Barcus Berry

Session engineer Taavi Moté at the SSL console in Larrabee Sound's Studio A

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R-e/p: What about the ending to a remixed single?

LS: Hmm . . . Yeah, the basic fade does get a little dated after a while! You know, I've done just about every ending that can be done. For example, the Stephanie Mills record, "Stand Back," will probably be out by the time this article is published. I remixed that from a version that was done by the producer, Nick Martinelli. He faded it out but, in the mix that I did, I came back into the chorus and then ended abruptly on a synthesizer sting. I think my version is more exciting, because it's like, bang! Bang! The song is over, and the DJ says the artist's name. There, I was thinking about radio, although the club version is the same.

So, I pay attention to all of that, but I do a lot of basic fades. You can destroy some material by trying to be "artsy" with it. It's not that difficult, especially after you've had a little experience: you get to know what the people want. And it's not always what *you* want. I take what other people say into account: I'm not the ears of the nation!

R-e/p: Do you take into account the possibility of a single crossing over into another chartable area?

LS: I used to, until "Oh Sheila" did what it did. That song went Number 1 on the *Billboard* Black charts, the Dance charts, and the Pop charts: just about every chart there is, "Oh Sheila" went Number One on it. So, to borrow a phrase from my boss, Jheryl Busby, "Crossover is just the sound of the cash register ringing." Which is to say that it has no *particular* sound.

You see, "Oh Sheila" was simply a good record that was given the opportunity to be exposed. And once it got played, people just loved it: it just snowballed. It's paved the way for that band. Now, we're going all out to get them a Platinum record. The album went Gold a couple of weeks ago [early January], and it's at 800,000 units now.

So, I don't even like to use the word "pop." I tell artists on the roster, "Don't think about crossover. Make the strongest record you can make, and let the company, and the public decide whether or not it goes pop." If you go into the studio with a preconceived notion that you want to make a pop record, you're going to alienate your core audience. The first thing we want to do is make sure that the base is covered, and the Black stations play this record. If they don't, Pop radio will *never* hear it.

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[Electronics]. It deals with phase and the time that the sound gets to the speaker. One of their claims is that, if you use the unit in the recording or mixing chain, you find yourself using less equalization — and I've found that to be true. On certain instruments, at certain levels, just dialing in a little of it really does enhance the track. I've used it on vocals, to add extra presence without adding level. You could say that it's a little like the Aphex [Aural Exciter] in the way that it's used. However, the Model 202R is not adding all that extra high-frequency energy: it works at a *cleaner* level. Most of the time, I place it in the mix on the same side as the source instrument is panned, and it adds 'clarity' to the track."

Enhancing the Width of a Stereo Sound Field

"Along with the outboard processing," Moté continues, "I'll also put up a couple of mikes and feed parts of the mix into the studio on an echo send. Then, I apply the sound of the room back into the mix, just to add a little acoustic material. By varying the position of the microphones and, possibly, compressing and limiting the room mikes — or even adding delay before the sound gets into the room — I can add dimension to the sound and give it more depth.

"A lot of things in the mix are done to give a wider stereo dimension. I try to create a lot of movement in the mix, so that things sound larger and wider: effects might start on the same side and stay there for a second or less, then be on the other side. Usually, I'll use an autopanner or make a mult of the track into two faders — one panned left, the other right — so that I can manipulate the faders with the [console-fader] automation, or use cut switches.

"My most recent discovery is the Cyclosonic FS-1 autopanner from SAN Musical Service Corporation. While demoing the unit, I found a setting where the sound kind of 'jumps' out behind you, then comes back into the mix. I sat there listening to a guitar jump out of the speakers and then jump back in: it was really phenomenal! I haven't incorporated that effect in a mix yet, because I don't know how it will translate to a record, but I plan to use it in the near future and see for myself.

"The point of all of this is to *enhance* the musical energy: to make it more dynamic, and make people want to get up and *move*. We take their favorite song and play off certain parts of it, giving it more meat, more bottom end, more dynamic range. You have so much liberty on a 12-inch: you can be more experimental. We also have the luxury to bring in other musicians and other devices —even record additional vocals — to carry out an idea that wasn't fully explored on the album version."

Are the differences between a dance-single remix and an LP mix extended to equalization, we queried? Moté indicated that this was indeed the case: "When I'm mixing, I think about the acoustical setting of the club environment. You don't want certain frequencies to be too strong or strident, or it could be uncomfortable, so I have to watch out for that. I wouldn't want it to be overly bright, for example, because at most dance clubs the music is played at a *very* high level. I *don't* want to get too strong in the 2- to 4- kHz region.

"But it's most important to be able to complement the song — that's number one. I listen to the lyric and try to understand the statement that's being made: for example, there might be a certain word or phrase that you can play off of in the remix, to bring out the meaning of the song. You can also create a mood with the outboard equipment, and the way that you write the mix into the [console] automation computer."

In the accompanying interview, Louil Silas had mentioned that, when extending a song, he worked in sections, and then had Moté cut tape to put them together. We asked the engineer to describe the process. "With the SSL console," he offers, "I'll generally get the mix up into a complete-pass situation with all the cuts written into the automation record, so that I have control over the sound; I'll also do some basic level rides if there are inconsistencies or low-level things on the master. Then, Louil and I will listen back to it, and pick up one section to work with, building up cuts and level moves. Sometimes, we'll work with the main body first, and get that developed so that it's really strong, then find the intro somewhere in the song.

"Of course, we begin by getting a mix that is close to what was on the original album cut: Louil likes to get to about where they were before, then make it *stronger*. It can be an awkward process, but I'll listen to the record or the two-track master, and try to figure out how it was done. Sometimes, I'll even call the engineer or producer to find out what they might have used! A lot of times, however, since we haven't had that luxury, I've had to sit back and listen carefully, and then try to match the track with something that I can pull out from my own usage of the equipment.

"The process relies on my analytical ear, coupled with good use of the outboard equipment. Who knows what was used to process a track? It could be quite complex! There've been a couple of times when Louil and I have done both the single version *and* the LP version, so we had the luxury of looking at the notes from before. But, most of the time, we have no other information than just going by ear."

Can we assume that the dance-single remixing process is an application where the engineer would appreciate transportability of data from one console automation system to

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"black" records to the other side. Take Klymaxx' "I Miss You:" before this album, Klymaxx sold about 12 units —I'm being facetious, but their sales record was *not* very good — and now they're close to shipping Gold. The New Edition story: because of the current album, *All For Love*, which was certified Gold this week and has only been out for two weeks, the first album will probably go double-platinum.

Patti LaBelle's "New Attitude" and the *Miami Vice* Theme [by Jan Hammer] are other examples. We're getting to be like a couple of other record companies, as far as both divisions just working good records. Of course, there's always going to be that division, because some records just don't work on other formats. But if we see a chance to make a record big, we go for it!

R-e/p: Do you ever become frustrated with the possibly artificial division that's made between "Black" and "Pop" music? Does it ever stand in your way?

LS: Well, there was one record that I knew would be big if it had the opportunity to be worked pop: "The Men All Pause," by Klymaxx. I think a lot of us in the company look back at that single, and feel that we made a mistake. I tried to get it reworked on the Pop side, but it was too late. The preconceived notion of a group, when they have no sales history, is "Why bother?"

Still, not every record is right for every radio station; the stations themselves aren't going for every audience segment. You might hear Tears For Fears on Black radio, but you're *not* going to hear Dire Straits or AC/DC. I

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mean, I know this "We Are The World" attitude — that it's all just music — but it's not really like that. You might hear "Oh Sheila" on Pop radio, but some R&B artists are so "funky" that Pop radio won't touch them. A lot of Rap records don't get Pop exposure, for instance. So, the division is still there. Some records should be played on every format, but there are also many that *shouldn't* be.

A lot of this business is just opinions, and some people justify their decisions on that basis, rather than going out and trying it. Like, I wanted "Digital Display" to be released before "Oh Sheila." Now, I don't know if it would have been a Number One Pop record, as "Oh Sheila" was: we'll never know. But had it been my decision, "Digital Display" would have been released first. I probably would have blown it, but that was *my* opinion. And a lot of it is personal opinion, but you try to go with the "pulse" of the market.

R-e/p: Why did you want "Digital Display" to be released before "Oh

Sheila"?

LS: You know how sometimes you just hear a record? Even from the demo, I heard that song as being a *great* record. To me, "Oh Sheila" just started and went to the end, and nothing ever happened in the middle. But I thought "Digital Display" was a

much more interesting track. When radio got the album, it was the first cut they went to after "Oh Sheila," so it proved to me that the song would have done real well. I also liked the subject matter, and it made a great video. We just finished the video last week up in the Bay Area, and it's got to be one of the videos of the year: a real "high-tech" video.

R-e/p: You liked "Digital Display" partly because it went through some changes. To me, we're touching on the question of song values: the structure, how the choruses and bridge fall. How do you relate to those things in the remix?

LS: With a song like "Oh Sheila," I try to incorporate things to break up the monotony. For that song, we used a tom fill to open the song and, then, right at the break, we did another tom fill that sent it somewhere else. Mixed that way, the song went on "levels," rather than just going from start to finish: it had a beginning, middle and end because of what we did in the remix.

R-e/p: But if a song already has a beginning, middle and end, how do you relate to that structure when you go into the studio for the remix?

LS: Are you asking whether I restructure the song? Because I have done that: I've put the middle at the beginning, or the second half in the first half. When I have the 24-track tapes, *I'm* the producer, and whoever produced the song originally has nothing more to say. Now, that could be called arrogance, but most of the producers feel good about it and don't look over my shoulder. When you're having a roll and you're successful, you get away with more! [Laughter] But if a song has good structure, I just try to

A Conversation with Taavi Moté — continued . . .

another? Would it be useful to have the fader-automation record from the LP mix to use as a starting point — assuming, of course, that he was remixing on a different brand of automated console from the type used during the original mixdown? "I think that would be wonderful," he concedes. "It would save a lot of time going through the song: sometimes, I've spent two or three hours just listening to the tape, making notes and writing [automated] cut switches. It's very tiresome, but has got to be done. It definitely would be an advantage to have all the automation data from the LP mix, stored in a form that was compatible with the SSL."

Does Moté have a particular preferences in monitor loudspeakers? "For close-field monitoring, I use the [Calibration Standard Instruments] MTMs — which I find to be *very* accurate — and I also use the standard Yamaha NS-10s with the tissue*. It's funny how the Yamahas became a standard, isn't it? The first time I heard them was at Westlake Audio [Los Angeles] in '79 or '80. I couldn't relate to them at the time: they seemed *real* bright. Now, of course, I like them. I hear the NS-1000s are pretty nice, too, although I haven't had the opportunity to work with them yet.

"Anyway, both during the mixing process, and also when checking the final mix, I listen on the big monitors; the close-fields, be they the Yamahas or the MTMs; and on Auratones, in stereo and in mono. I'll even use the small [cuing] speaker in the Studer A-80 or the one in the [Ampex] ATR-100; I also take a real 'el-cheapo' mono cassette deck with me as another reference. If I think I've covered everything on all those speakers, and nothing is drastically bothering me, then I know I'm safe anywhere!

"It's important to monitor at low levels sometimes, as well. Louil works at high volumes, and I have to have a reference point if my ear gets a little fatigued, to make sure that everything is still within the framework at a lower level. High levels can trick you after awhile: the ear is a very complex, sensitive part of the body, and it's *definitely* going to change its curve after exposure to sound at an excessive level. After maybe a short ear break, I reference on small and medium speakers — even if we're talking about a mix that's going to be played at high volumes on a dance floor. If everything's there, then in a club in the Middle East or Jamaica, which may have a very different sound system, we still have a mix that'll work." □□□

*****Editorial note:** The interested reader is referred to a fascinating article by Bob Hodas, to be found elsewhere in this issue, that sheds light on the "industry wisdom" that placing tissue paper over the NS-10M's tweeters will reduce the perceived HF content of a mix. As the article reports, after extensive tests Hodas attributes the affect *not* to simple HF absorption, but rather to a complex *comb-filtering* process — ML.

accentuate what's there, and make it sound stronger.

R-e/p: Do you refer back to the album cut, and attempt to remain consistent with that sound?

LS: Oh, definitely! When we first come into the studio, we get the song sounding exactly like it did on the album, and then go from there. You can always make it sound *better*, but first you have to make it sound the *same*. And I can't think of one time where somebody would rather play the LP version than the remix. Of course, I'd probably say that even if they did! [Laughter]

R-e/p: When working on a remix for an established artist — say, a song from a second or third album — do you listen at their previous albums, and try to be consistent with their earlier work?

LS: I did that when I first started doing remixes, but now I just go for it. Take Stephanie Mills' "Stand Back," for instance. I felt that this was the 1985 version of her, in which case the prior material doesn't really relate. And I certainly *don't* want any more unsuccessful albums like the last two she had on Polygram. I want this album to be her biggest: it has to go Platinum, because she's had Gold records. So, I went with what was recorded and just dealt with that. We have a *very* strong track, and she gave us an excellent vocal. I think that this single is going to surprise a lot of people, and make her prior record company green with envy!

R-e/p: I would assume that you pay particular attention to where the artist and audience are now?

LS: That's right. You know, if I have a chance to re-record a lead vocal, I'll do it, even if the artist thinks that the original vocal was great when they did it four months ago. I like to get their attitude on the night of the remix: their thoughts have got to be different than they were four months ago. They might have lived with the track, and now want to give a different interpretation. You know, I usually do a remix about five weeks before the record comes out, so if I can get some feeling of immediacy right then, it can make the remix even fresher.

R-e/p: And the artist's ears may have changed a little bit, too — from what they've heard on the radio, for instance.

LS: Exactly. You know, a lot of this is a product of what you've just heard. I hear a *lot* of material on the radio; I go to clubs all the time; and I'm always listening to see what people react to. If

somebody else has done something that makes people go crazy, I follow the way. I'm not saying that I steal, or anything like that, but I want a *successful* record and I'll do anything — without blatantly stealing somebody else's idea — to get it!

I've heard a lot of my elements on other people's productions, too, after all. It's a form of flattery. I know that guys who are remixing listen to things that I've done before they go into the studio, and it's good to do

that. As long as you have a successful record, who cares?

R-e/p: What do you look for in a new artist?

LS: Something that grabs me. Since I've taken on the A&R position with MCA, I've only signed one new artist to the roster, although I've listened to about 15 million tapes! Now, I'm signing a couple of groups that have been on other labels, but only one new artist: he's a male vocalist named Giorgio — out of Minnesota, but *not* of the "Prince school." I signed him to an LP deal, and his album will probably come out in March or April.

When I first heard Giorgio, I was sitting in my office one night at about 11 o'clock, listening to tapes. He had submitted a video with two songs on it; I put it on, and I watched that video over and over until five in the morning. I couldn't believe it! He's just great. Vocally, he is a little like Peabo Bryson: he's a great balladeer.

What I heard in his voice was just a raw, "sensual" appeal. I thought he was a guy whose voice ladies would love, and the guys wouldn't be threatened by it. He's not a singer in the "Teddy Pendergrass" mold, where he's going to be demanding that your woman "move over here," you know! He's going to be a singer who sets

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your lady up, so when you take her home from the concert, she's ready.

The group I'm signing is called Body — three sisters from Detroit. They worked with Stevie Wonder for what seemed like years but, for some reason, never got an opportunity to record. A road manager of New Edition brought them to my attention, and they came to my office and played me a demo tape. I just knew that these girls deserved their shot.

Basically, I'm looking for something new and exciting. Right now, if there is any young man out there that is 16 years old, but has the voice of a 30-year-old man, make sure you get your tape to me! I'm looking for an MCA version of Johnny Gil, who is on Atlantic. Johnny must be 18 or 19 now, but he sounds like he's 40; he has great "command" of his voice. I hate to talk about artists on other labels, but Johnny's lucky. He came from the church, so he has what he's learned in church combined with what he's now learning in show business. I think he's going to be a star for years to come.

I feel the same about our guy in New Edition, Ralph [Tresvant], and I also feel the same about Giorgio: once he gets out there, people are going to find an attractiveness in his voice that they haven't heard in years. When I signed him, I took Jheryl Busby to Minnesota for the Black Music Awards in September, and Giorgio opened the show. He just blew Jheryl and the audience away. Tell your readers to hold onto this article, and we'll see in a year where Giorgio is, and where I am! [Laughter]

R-e/p: In spite of the fact that you were looking at a video, it seemed to be the sound of Giorgio's voice that grabbed your attention?

LS: Yes, it was, but the video gave me the opportunity to see him. I love it now that we're in the Video Age, where you can see what the artist looks like. That's the next thing that you want to know if you like the voice, and sometimes they don't look like anything!

So much of this thing is how well the artist is prepared. This is not Motown anymore, where you have a school grooming people to be "stars." They have to be pretty much ready when they send their material to the record company. A lot of the tapes that I get in, I'm glad that there are no videos, because I can tell that they're not ready.

R-e/p: Isn't an artist's material also very important?

LS: Of course. You know, I've learned that you can do this thing two ways. You can rush through a project — and think you're looking for good songs, but just accept anything — or you can keep on pounding until you find *those* songs. You've got to be able to say "no" a lot, even to things that you might love but that you know other people won't.

I think the Stephanie Mills album [*Stephanie Mills*] is a good indication of where I am as far as looking for material, and putting it with an artist, is concerned. Of the eight songs that are on the LP, I was responsible for submitting five.

I was looking for something special for Stephanie, and we found some songs from guys who write more than superficial lyrics. I mean, there are many ways to write about romance, tragedy, whatever; but there's always a superficial way, and then there's a transcendental way. Rod Temperton is coming from that space — he can take a subject and, by his lyric content, really make that song special.

There's a songwriter coming up by the name of Raymond Jones, and he has a song on this album called "Rising Desire." It was produced by George Duke and, lyrically, you can't beat it. Stephanie just kills it!

So, even if the groove is great, if the song's not *saying* anything lyrically, it has to go. A lot of people think that this "Everything has to be a great song" attitude is going to kill me, but I truly believe that, depending on the artist, a *lot* of the songs should be special. Patti LaBelle needs a special lyric — she can't just sing anything! Luther Vandross also needs a special lyric. Certain artists need more than an *ordinary* song.

R-e/p: Isn't that because they're more than an "ordinary" artist?

LS: That's a good question, and how can I weasel my way out of it? [Laughter] You get in trouble over what's an "ordinary artist"! In the case of Patti LaBelle, Luther Vandross, Stephanie Mills, Whitney Houston or Freddy Jackson, they are exceptional. I don't want to get in trouble with anybody else but, with an artist like that, you're going to examine and examine, to make sure that this is the *right* song. These songs for Stephanie just clicked. But a song that one artist turns down may be right for another one on your roster. Stephanie turned down some songs that I thought were right for her and, since getting to know her better, I now know what to look for. She is my success story: that's why I'm hoping her album will be big. I did a heck of a song search on that album!

R-e/p: One last question. Since you have the advantage of hindsight, you're in an enviable position during the remix stage, because you don't have to make that first shot at taking a song to the stage where it's got a chance of becoming a hit. Would you like to produce an artist from the ground up?

LS: Oh, yeah. I have a production deal with MCA, and I'm getting into production now through co-producing — just getting my feet wet. An artist that we just signed — a young lady out of Philadelphia named Janis McClain — is going to be my first co-production, with a gentleman by the name of Ron Kersey. And I'll be producing Giorgio myself, from the ground up.

So, production is something that I definitely want to get into, but first I want to be the *best* A&R director that I can be; I want to sign some artists that have double-Platinum albums, and make my mark as the guy that everybody wants to bring their great artists to.

The remixing thing is just a phase. First of all, the schedule is just a monster: it's broken up a couple of relationships, because I just don't have time. But you're out for self-fulfillment, to be as successful as you can. You can't let anything or anyone stand in your way. I would rather be a success, and then deal with the other thing.

I'm going to produce, and I hope to be a good producer. I think that with the team Taavi and I have now, we have the stuff to make some *great* records. I've been approached by people from other labels, but while my deal with MCA is flexible, it doesn't call for me to produce people on other labels! [Laughter] You know, "Hey, Louil, I saw you're Number One on Luther! You should be working for CBS!" But, right now, I want to take it slow. I'm enjoying A&R, man. I'm using all that I'm learning now, and I have some ideas that I think ultimately will help people sell some records. ∎∎∎

Touring concert sound companies have helped to bring about the development of some significantly new manufactured products, and system operating techniques for use in live performance. Wireless microphone systems, multiple-output portable mixing consoles, kilowatt power amplifiers, and full-bandwidth stage monitor enclosures — all these innovations, and others, were brought into existence with the help of touring sound companies to satisfy specific needs during the past 20 years. The existence of these and other products is a testament to the pioneering efforts of a wide range of concert sound companies.

Mixing consoles, power amplifiers and loudspeaker enclosures are among the first major components in a concert sound system to be designed and built, or purchased. In the earlier days of touring concert sound, the major equipment manufacturers did not offer products that were suitable to the needs of the rapid growing tour industry. As a result, sound companies often had to develop their own proprietary devices to solve the challenges presented by highly amplified popular music.

Today, audio equipment manufacturing techniques have progressed to the point that most touring sound companies are able to satisfy their equipment needs by choosing from products available on the open market. There are exceptions, of course, but a survey of sound equipment in use with touring shows in auditoriums, arenas and stadiums shows that an ever-increasing number of performances are relying on consoles, amplifiers and loudspeaker enclosures that are not the proprietary design of a particular concert sound company, but rather products available from well-known commercial equipment suppliers. It should be stressed here that design input and product development assistance from major touring sound companies has, without a shadow of a doubt, helped to bring these products to the marketplace, and that the touring sound industry has brought about an overall improvement in pro-sound products.

Ultra Sound: Company Origins

A highly visible example of a touring concert firm that has chosen to work with equipment manufacturers to develop new products for live performance use is Ultra Sound, Inc., based in Larkspur, CA. Working within the past decade, partners Don Pearson and Howard Danchik have attempted to take a studio-quality approach to concert sound. The company's Gamble/Crest/Meyer sound

HIGH TECHNOLOGY ON THE ROAD

The Grateful Dead in Performance at the Brendan J. Byrne Arena, Meadowlands, Utilizing an Ultra Sound System

by David Scheirman

ON THE ROAD WITH THE GRATEFUL DEAD

reinforcement systems serve a select client group, including such artists as Crystal Gayle, George Thorogood and the Delaware Destroyers, and the Grateful Dead.

"Before such products as the Gamble consoles, Crest amplifiers and Meyer loudspeaker enclosures were available, we used to do our own research and development," explains Ultra Sound president Don Pearson (*pictured left*). "In the touring sound business, we started by having to learn to build a project or fix an amplifier overnight in a hotel room with a minimum of test gear. Now, we can build or service almost anything in audio.

"I am probably more of a scientist than a businessman; I love to invent things that will solve sound system problems. Lately, I have had to spend more time with business than inventing, and that takes the fun out of it. Being able to find manufacturers that we can work with — who are able to construct devices that stand up to our rigorous demands — has been very beneficial to Ultra Sound."

The company's current sound systems are built around components from three manufacturing firms whose inception and growth have roughly paralleled that of Ultra Sound: Jim Gamble Associates, Crest Audio, and Meyer Sound Laboratories.

"A concert sound system is no longer just a jumbled pile of parts," notes Pearson. "Each component in the signal path must be matched and work well together. Many of the audio products advertised today do not even meet their *own* published performance specifications, so it is difficult to try to build a system around that sort of gear; you'll have some pieces that just don't fit. We looked for companies that knew what *we* needed because of their first-hand experiences in this field. They have supported us, and we have supported them. What we have come up with are some sound systems that I think will perform better and more reliably, on a consistent nightly basis, than anything else available to rental customers."

System Components

The typical Ultra Sound system has a relatively simple signal flowpath, due in part to the use of Meyer signal processors that do away with the need
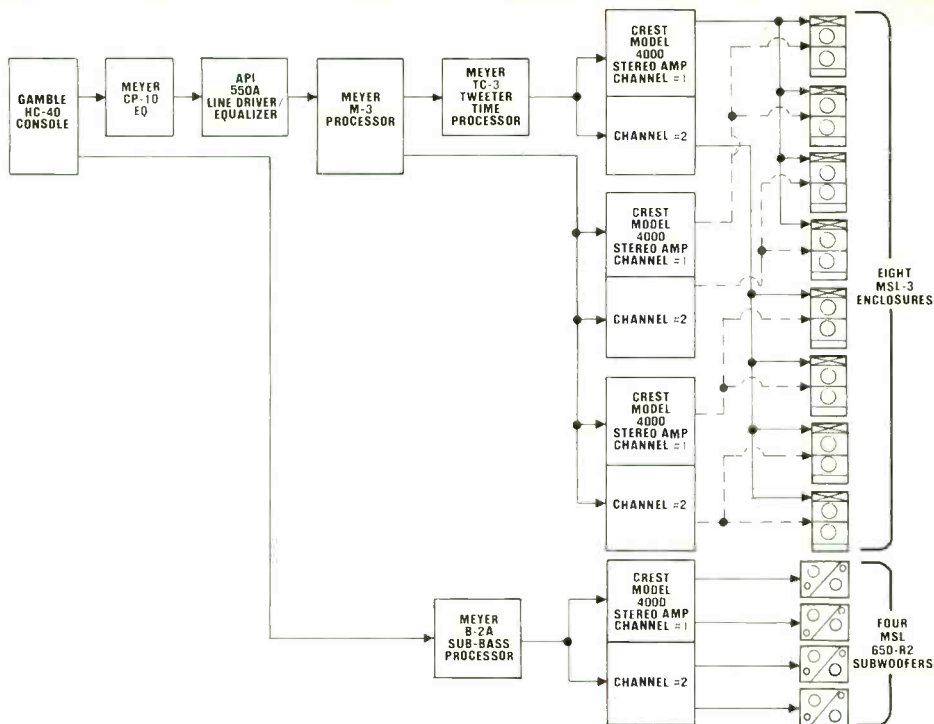
Figure 1: Main Signal Flowpath of Ultra Sound Concert System.

for separate electronic crossovers, passband limiters, and loudspeaker protective devices (Figure 1). Since the Gamble HC-40-24 console offers 16 output busses in any configuration, adjustments of left and right subwoofer levels can be made at the console, in addition to the stereo mains. Rear- and side-fill levels are set at the amplifier racks.

"It has taken some concert sound technicians a bit of getting used to, having no control of the various crossover levels at the mixing console area," explains Ultra Sound technician David Robb (*pictured right*), who is responsible for operating and maintaining the company's East Coast warehouse and shop in Woodstock, New York. "Those of us who have worked closely with this system have come to see that John Meyer's concepts offer some definite advantages. The overall consistency of the sound of the system from night to night is very, very good. Regardless of who the system engineer is, what type of musical show we are doing, or what sort of location we are in, the Meyer equipment lets us achieve a better standard of quality."

Gamble HC-Series Consoles

Jim Gamble's custom, portable mixing consoles have become a favorite of

many concert soundmixers. Features such as transformerless circuitry, onboard parametric output EQ, and programmable muting have made the designer's products popular with many independent live-sound engineers. [See: "On the Road Again . . . the Willie Nelson and Family Tour," *R-e/p* April, 1983 issue — *Editor.*]

Ultra Sound worked closely with Jim Gamble Associates during development of these live-performance consoles, and currently offers three sets (house and monitor) for rental use. The consoles are equipped with front-access patchpoints and onboard spectrum analyzers (Figure 2).

"I started building these consoles because I saw that a market existed," states Jim Gamble. "Existing electronics firms that try to cater to the touring sound business have to build in a lot of compromises in their products to keep the costs down. They can't sell enough units to justify the research and development cost for products that have both the needed features *and* the necessary audio quality for discriminating firms like Ultra Sound. I build boards for sound companies and bands that are into good sound, not just making money."

Gamble, whose own career as a live soundmixer began around the time of the 1967 Monterey Pop Festival, handcrafts the consoles at his shop in Tahoe City, CA.

Crest Amplification

Power amplifier design for live performance audio use has improved

Figure 2: (Shown left) Jim Gamble Associates' HC-40-24 mixing console as supplied to the Grateful Dead by Ultra Sound. Figure 3: (Shown right) Ultra Sound packages its Crest amplifiers in two different types of racks. Four Model 4000 units are housed in compact racks, while larger systems use five Model 3500 and two Model 5000 amplifiers per rack.

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greatly within the past decade. The days of delayed concerts due to overheated, overburdened low-power amplifiers are practically gone, thanks to a new generation of amplifiers now available to sound companies. A variety of firms now market powerful, reliable amplifier packages that provide high-wattage "building blocks" for touring concert sound systems; such units were only a dream a decade ago.

While some companies have recently introduced products based on "new" technologies — such as the magnetic field and the digital conversion amplifiers — Crest Audio has specialized in the design and development of optimized Class AB power amplifiers with well-regulated conventional power supplies. Hardly five-years old, the company's product line, which contains such devices as the Model 2501A, 3000/3001 and 4000/4001, has been judged to be one of the most reliable for use under the extreme conditions faced in touring concert sound.

The company claims an FTC twoohm stereo power rating of 800 Watts per channel for the Model 4000. Due to the greater amount of iron found in the power transformer, such amplifiers are larger and heavier than some of the new, unconventional products on the market. However, many concert sound system designers, including Ultra Sound's Don Pearson, feel that this is still the best all-around type of amplifier for use in massive concert sound systems, particularly where very low frequencies are concerned. "All the different amplifiers work, and they all can get loud, the different is which ones will sound 'sweet' at concert levels," comments

Pearson. "In both our electrical bench tests and subjective listening tests, the Crest units seemed to be best suited to our needs."

Ultra Sound has chosen an input sensitivity "standard" for the amplifiers in their systems so that every power unit in each system is exactly calibrated. "Every brand of amp has a slightly different sensitivity rating, and the variance from unit to unit on some manufacturers' products can be significant," Pearson notes. "We have chosen to have one volt of input produce 100 Watts of output into eight ohms — that's a gain of 27 dB. That way, all parts of a large system are responding to the input signal in the same way and, when huge systems are put together from smaller systems, the result is *very* consistent."

Crest Audio president, John Lee, stresses that touring sound companies have really helped both the audio equipment industry, and the concertsound business. "This trend of people demanding better live sound has done a lot to give audiences a better experience, because the sound companies continually have had to keep improving their systems," Lee notes. "Building a commercial power amplifier is something like putting together a Formula I racing car: there are many different design approaches to take, but the whole package must do what it does to the *best* of its ability, without failure. The large sound system operators are just like the racing car drivers . . . they know, better than anyone else, whether things are working properly or not."

Figure 4: 112 Meyer MSL-3 enclosures were suspended for the Grateful Dead's two-night performance at the Meadowlands Arena in November, 1985 — reportedly the largest indoor Meyer speaker array ever assembled.

Figure 5: Front of Meyer MSL-3s.

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Lee feels that companies utilizing contemporary amplifier technologies may have sacrificed audio quality for increased efficiency and lighter weight. "Some of the new amplifier products are more efficient than conventional designs, but at what price to sound quality?" he queries. "The biggest criticism of some of the lightweight amps is the sound in the low end."

What an amplifier (or any other audio product, for that matter) "sounds" like is very subjective. Oftentimes, side-by-side listening tests of different amplifiers have been hampered by the minor inconvenience of setting up such a test with its different connectors, and so forth. To facilitate the quick and easy comparison of different power amplifiers, Crest recently developed a six-channel "A-B" box for use by sound companies and dealers.

Ultra Sound has packaged its Crest amplifiers in two different types of racks (Figure 3). Compact racks are available, each housing four Model 4000 units (rated at 800W per channel into two ohms, FTC). Older, larger racks house five Crest Model 3500 and two Model 5000s.

Meyer Loudspeaker System

Since its inception in 1979, John Meyer and Meyer Sound Laboratories have worked closely with Ultra Sound on the development of the current MSL product line. Starting with the UltraMonitor™, Meyer began to attempt to create loudspeaker enclosures that were truly phase coherent. One of the company's early, successful products was the UPA-1A, a compact, full-bandwidth, arrayable enclosure that featured John Meyer's M-1 Control Electronics Unit. This latter device contains an electronic crossover with frequency- and phase-response alignment circuitry. Meyer's exclusive SpeakerSense™ circuitry protects the loudspeaker components from damage due to overheating under high-power operating conditions.

"What we are trying to do is more than just putting out products for sale," Meyer explains. "This product line is the result of many years of research into the needs of live performance groups for both touring and installed systems. We have come up with system 'building blocks' that can be combined into a truly high-powered system. Our loudspeakers do a lot more shows around the planet in a four-, eight- or 12-cabinet system configuration. However, when they are combined into very large arrays, such

as Ultra Sound has been doing for the Grateful Dead, the true potential of these cabinets is realized. Both the UPA's and the larger MSL-3 enclosures provide a measurable full-frequency response of ± 3 dB across their rated polar patterns. Just for comparison, some heavily-advertised loudspeaker systems measure ± 20 dB across *their* polar patterns."

In November 1985, Ultra Sound assembled what has been described as the largest indoor Meyer loudspeaker array ever for the Grateful Dead's two-night appearance at Brendan J. Byrne Arena in the Meadowlands, New Jersey. A total of 112 MSL-3 enclosures and 30 650-R2 subwoofers were combined into what was perhaps the most massive amount of acoustical energy ever used for an indoor live performance (Figure 4).

"One hundred and twelve cabinets . . . that's quite a statement," Meyer offers. "We don't do well if people are looking for a large visual display. Then, it's too expensive, and the extra gear is not really needed to get the necessary sound. The system assembled for the Meadowlands show would have looked *huge,* if a traditional sound system had been used to get the same acoustic output."

The MSL-3 enclosure houses two proprietary 12-inch, low-frequency cone drivers in a horn-loaded vented enclosure. A single high frequency driver is mounted on a 70-degree horn, and four-unit very-high-frequency array is arced in the cabinet beneath the horn (Figure 5). The trapezoidally-shaped cabinets can be combined easily into compact stacked or flying arrays (Figure 6).

Each 650-R2 subwoofer houses a

Figure 6: The trapezoid-shaped MSL-3 enclosures are easily combined into compact arrays.
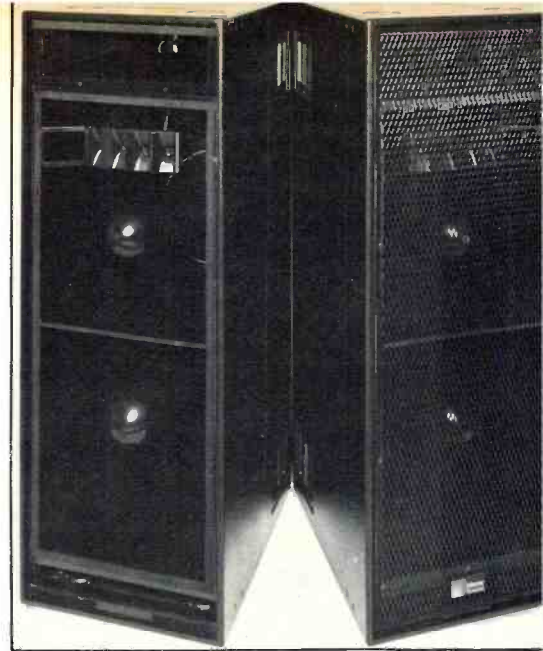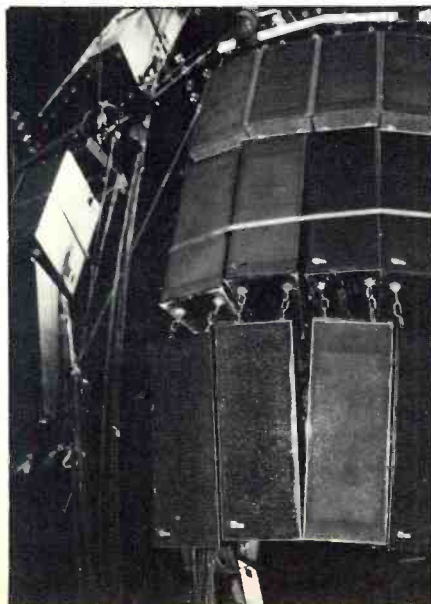
pair of 18-inch low-frequency loudspeakers in a heavily-braced, ported box. The latest development in a line of subwoofers that started with the 650EM subwoofer cabinet, the -R2 was developed originally in conjunction with the production team for the film *Apocalypse Now,* to accurately reproduce the sounds of helicopter gunships and Howitzer mortars (Figure 7). Besides Meyer Sound Laboratories' aforementioned sound reinforcement products, Ultra Sound also supplies stage monitor systems that feature the UM-1A speaker enclosures (Figure 8).

Sound for the Grateful Dead

For 20 years, the Grateful Dead has been known as one of the most technically demanding and sound-quality conscious musical groups to ever take a show on the road. In the past this San Francisco Bay-area band has attracted a great deal of interest in the media, due in part to the never-ending quest for better live sound by Dan Healy, the group's live sound engineer.

"After we experimented with the 'Wall Of Sound' in 1973 and 1974, we

realized that it just wasn't practical for the band to own such a system and carry it around," recalls Healy (*pictured left*). "We went through a period of several years, searching for a concert sound company that was forward-thinking enough in their design philosophies and operations to give us what we needed. For about five years now, we have worked exclusively with Ultra Sound; it is the best working

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relationship I have ever had with a PA company."

The sound system assembled for the Grateful Dead's shows at the Meadowlands Arena comprised Ultra Sound's regular touring rig for the band (64 MSL-3s and 20 650-R2s) along with supplementary gear supplied from Ultra Sound's East Coast warehouse facility. Additionally, Meyer cabinetry was subcontracted through two area companies: Audio Support and Capital Sound.

"Most of the major sound companies want to have a certain identity," explained Ultra Sound president, Don Pearson. "Sometimes they think that identity has to come from their own particular speaker boxes or whatever. They have perhaps been hesitant to accept loudspeaker systems from a manufacturer, such as Meyer, to which their competition could also have access. We have seen that we definitely have our *own* identity, even though we use a speaker system that is available all over the world. How does a company set up their gain structure? How do they fly the system? What sort of events are they doing? Every sound company has an

identity, no matter what speaker system they use. For us, the Gamble/Crest/Meyer combination is the hardware side of things; design, operations and servicing the accounts are a big part of the picture as well."

Stage Monitors

The Grateful Dead's monitor mixer, Harry Popick, operates a Gamble SC-40 stage console.

"It's a very simple system," explains Ultra Sound technician Mike Brady (*pictured right*). "Each performer takes one mix output. The Gamble console has parametric EQ on each output, so outboard equalization is hardly ever required. We have some Meyer CP-10 [parametric] equalizers in line for when things get really drastic, like when there are extreme temperature changes on stage before showtime that affect the sound (Figure 9)."

An additional mix is provided for backstage guests. A variety of interesting technical fixes have also been incorporated into the stage monitor system, including a dbx Series 900 rack filled with noise gates for the
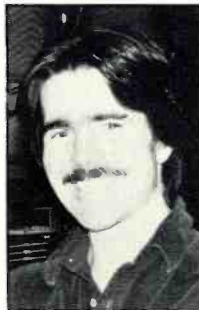
Figure 7: 650-R2 subwoofers were stacked five high in rows of three, for a total of 30 boxes in use at the Meadowlands performance.

vocal mikes, and which are voltage-triggered by elements tucked beneath rubber footpads.

Power Distribution System

One of Ultra Sound's technological standouts is in the area of power distribution systems. A portable electrical power system supplied with the company's large systems is more sophisticated than the equipment installed in most large performance venues, as can be seen from Figure 10, an Apple Macintosh-generated diagram courtesy of Don Pearson.

The main distro rack contains a 400-amp main breaker for three-phase service; feeders are connected here from the main electrical service panel (Figure 11). Sub-distribution panels are provided for the main power amplifiers, plus the stage gear and monitor system. All consoles, signal processing devices and other electronic gear are provided with a motorized, regu-

Figure 8: Ultra Sound stage monitor systems feature MSL UM-1A speaker enclosures.

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Figure 9: Stage Monitor System.

Figure 10: Power Distribution Block Diagram.

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lated power supply that will provide clean, balanced voltage nearly anywhere in the world.

"One of the most overlooked aspects of sound reinforcement system design is the power distro system," Pearson stresses. "We feel it is imperative to be able to balance the load that is placed upon the system. A musical performance is very dynamic. When the power load is distributed equally over the service legs that are available, the end result is a system that is *much* more stable. When the current drop varies across the different legs during the performance, the stability of the power amplifiers is affected. We find that the best thing is to first look for the most stable, reliable, good-sounding amplifiers you can find. Then, supply them with a very clean, balanced power source. The improvement in total system performance *is* audible."

Flying the Speaker Array

For the Grateful Dead shows at the Meadowlands Arena, the 112 MSL-3 enclosures were suspended in a unique flying system array that had been carefully calculated by the sound crew after examining architectural drawings of the building. This particular arena, with approximately 20,000 seats, is characteristically one of the largest enclosed volumes of air of any similar facility in the nation, due to its high ceiling and approximately square building shape (as opposed to
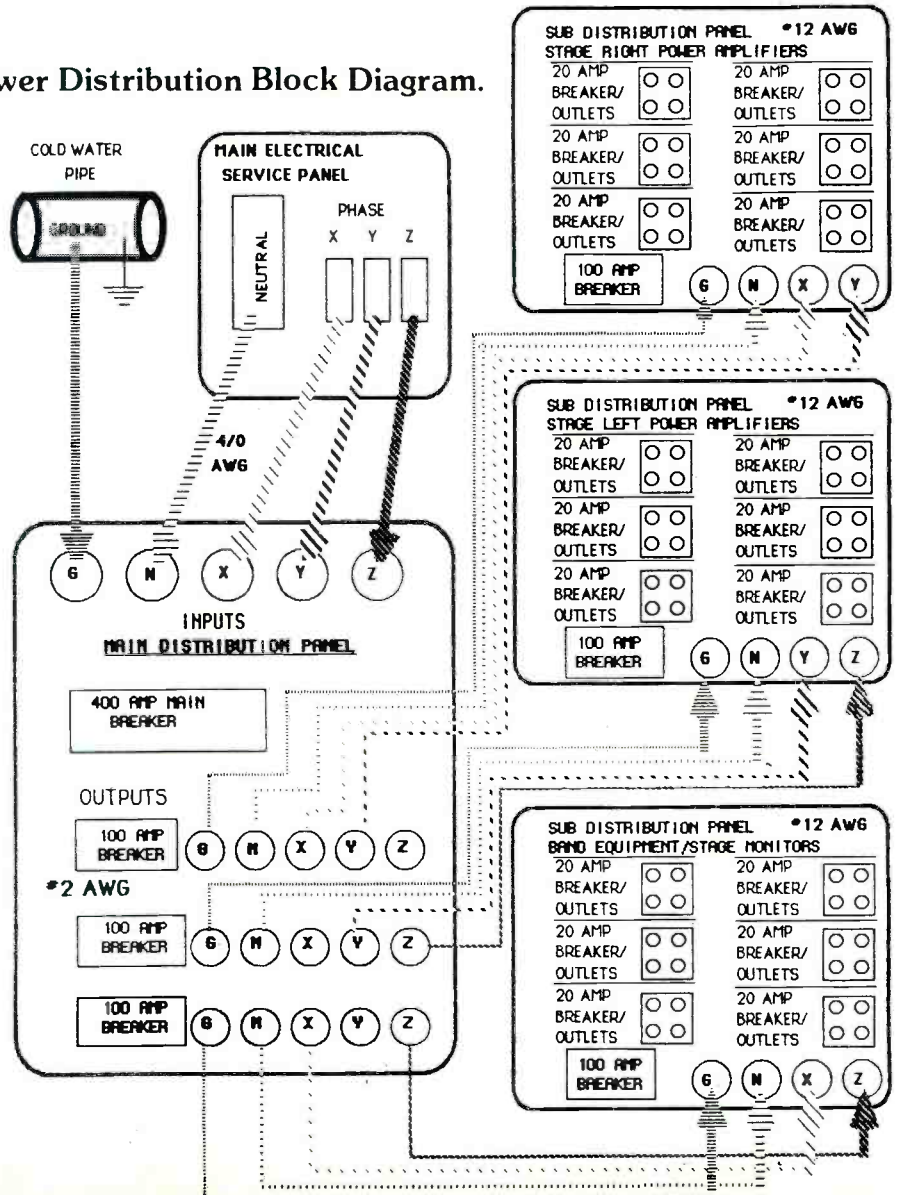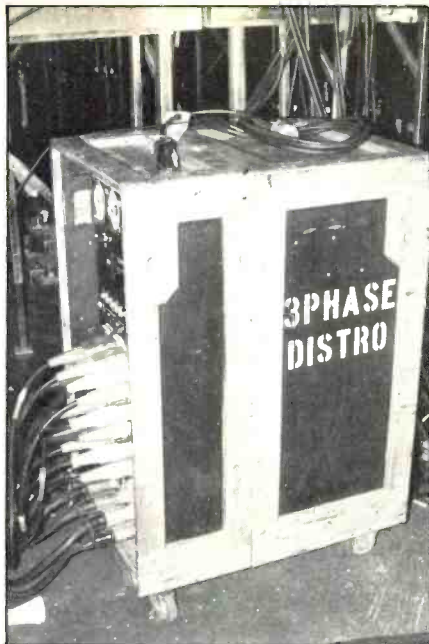
Figure 11: The main power distro rack offers three-phase, 400-amp service.

long oval facilities such as Madison Square Garden).

"For the shows here at the Meadow-lands, we have made consideration for elevation and distance from the audience of each bank of speakers," Ultra Sound technician Dave Robb points out. "The cabinets are coupled most tightly on the top rows, where the speakers are the greatest distance from the audience members. For the main floor area, we have set up what we call 'Power Alley' for mixer Dan Healy. That is an area approximately 20 to 30 feet behind the mixing console, where the intersection of the axes of the left and right main arrays takes place (Figure 12)."

An inner hang of 20 MSL-3s per side was made with seven one-ton chain motors; another seven motors handled an outside hang of 16 cabinets per side. A side-hang array used two motors for 12 cabinets for left and right, while an additional eight boxes per side were suspended by a pair of motors in each rear corner to cover seats behind the stage (Figure 13).

The side-hang arrays were widely spaced, to reduce that part of the system's impact on the close seating areas, and to obtain a wider coverage spread. The subwoofer stacks were wired out of phase with the MSL-3 arrays to compensate for path length differences.

"The distance of the wavelength at the crossover frequency is so long that there would be no audible phasing problems if the subwoofers were located approximately beneath the flying arrays," explains technician Dave Robb. "We run them out of phase because the main arrays are several feet out over the audience

Figure 12: The main floor seating area was covered with a specially-aimed speaker array that created a zone known to the sound crew as "Power Alley."

Text inside Figure 12:

STAGE AREA

LEFT ARRAYS

RIGHT ARRAYS

CONSOLE

SHADED AREA: "POWER ALLEY." AN AREA OF HIGHLY FOCUSED SOUND ENERGY WHERE SEATS ARE PURCHASED BY AUDIOPHILES, AMATEUR TAPE RECORDISTS AND HIGH-LEVEL MUSIC FANS

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from the stage line where the subwoofers are placed."

(Note: The wavelength of a 100-Hz signal is approximately 11 feet, and a physical displacement of the subwoofer system with respect to the main arrays of 5.5 or 16.5 feet — 0.5 and 1.5 times the wavelength — will produce a serious dip in frequency response at 100 Hz, for example. Reversing the phase of the subwoofer system with respect to the main system reduces this effect.)

House Mix Position

Ultra Sound supplied Dan Healy with six Meyer CP-10 Complimentary Phase parametric equalizers, giving a total of 30 filters available per side. A Bruel & Kjaer Model 2032 computerized analyzer is used for observing frequency and phase response of the system at the console. The musical performance provides the test signal for the Fast Fourier Transform (FFT) device (Figure 14).

"The CP-10 project grew out of measurements that we took several years ago during some Grateful Dead performances," explains John Meyer. "As I recall, we were about 130 feet from the large system outdoors at San Jose, CA, with a Hewlett-Packard analyzer. There was a huge peak at about 130 Hz, maybe 10 dB or so. As we worked with the EQ of the system, we began to realize that the equalizers in use had filters that were so interactive as to be actually *adding* to the problem. Whenever more than 6 dB of cut was introduced at a particular frequency, it began to affect the adjacent filters' phase response. The CP-10 has been developed to solve that problem (Figure 15)."

Dan Healy and Don Pearson are using the B&K Model 2032 analyzer to implement John Meyer's SIM™ technique (Source Independent Measurement), which is a method of analyzing sound-system phase and frequency response during the concert itself. [See a feature article detailing SIM applications during the Golden Nugget Casino design; April 1985 issue of *R-e/p — Editor.*]

"Before we developed SIM, everybody else was using some sort of impulse testing, which of course is not possible during a show," Meyer continues. "Now, we are getting closer and closer to being able to actually do the measurement with the music of the performance, and see the display in real time. We are only limited by the speed of the computers in the analyzing devices. The B&K is a very good tool; a good intermediate step on the

REAR-HANG:
8 MSL-3 CABINETS

STAGE AREA

SIDE-HANG:
12 MSL-3 CABINETS

SUBWOOFER STACK:
15 650-R2 ENCLOSURES

OUTER HANG:
16 MSL-3 CABINETS

INNER HANG:
20 MSL-3 CABINETS

Figure 13: Top View of the Stage-Right Flying Array.

way to that goal." (For a further explanation of the FFT Analyzer, see accompanying sidebar.)

Ultra Sound supplies Dan Healy with a wide array of signal processing and special effects devices. Lexicon PCM-42 and Model 200 digital delay and reverb units, a Roland SDE-3000, a DeltaLab Echotron, a dbx Model 500 subharmonic synthesizer, and a Klark-Teknik DN780 stereo digital reverb are all housed in the main electronics rack at the console.

A dbx Model 166 Overeasy limiter is provided for use with occasional guest acts. "This system will get loud very quickly," Pearson notes. "Inexperienced engineers often found with opening acts, who are not familiar with Meyer systems, get themselves in trouble sometimes by pushing the sound pressure levels way too high before they even realize it. The customary subharmonic distortion that some people associate with volume in traditional speaker systems is not present in the Meyer gear."

To make things go more smoothly at the house board, Pearson is continually indulging his inventive self with custom electronics projects. A recent addition to the system is an electronic patchbay, based on an IMS (Integrated Media Systems) switching device, to handle the various pre-

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DYNAMIC REALTIME CONTROL OF SYSTEM EQ DURING LIVE PERFORMANCE

Fast Fourier Transform (FFT) devices are used in a variety of scientific measurement applications, ranging from fluid dynamics to naval architecture. Manufacturers such as Hewlett-Packard, Panasonic and Bruel & Kjaer recently have begun to increase their research and development efforts for audio-related applications in the measurement and correction of live-sound systems.

Dual-channel FFTs, such as the B&K Model 2032, are easily patched into sound reinforcement systems for measuring both the electrical output, and the performance of the system in an acoustical environment. To accomplish this feat, one channel of the FFT receives information from the mixing console output, while the other channel is fed with signal information from a calibrated microphone located in the performance area. A delay line inserted in the console output feed is precisely adjusted to compensate for the distance between the speaker system and measurement mike, which is located typically at the house-mix position. The FFT analyzer then enables the sound-system technician to readily spot anomalies in the speaker/room frequency response. As corrective changes are made via the system's equalization device(s), data-averaging is used to indicate the effect that such changes are having within the concert hall.

Because third octave graphic equalizers commonly seen in many sound systems do not allow sufficiently precise adjustments to the system when dealing with very narrow-bandwidth room resonances, parametric equalizers with non-interactive filters work best with this FFT procedure.

Many low-frequency rumble and mid-range ringing problems with large sound systems operating in acoustically-poor rooms are caused by exceptionally narrow-band frequency peaks that do not register accurately on the third octave real-time analyzers often used today. For this reason, when combined with tunable narrow-bandwidth notch filters, the FFT analyzer represents a significantly more useful tool than conventional methods, such as pink noise, EQs, or analyzers. (It is also worth noting that the actual *live* music of the rehearsal or performance can be used as the excitation program material to measure the sound system's interaction with the acoustic field.)
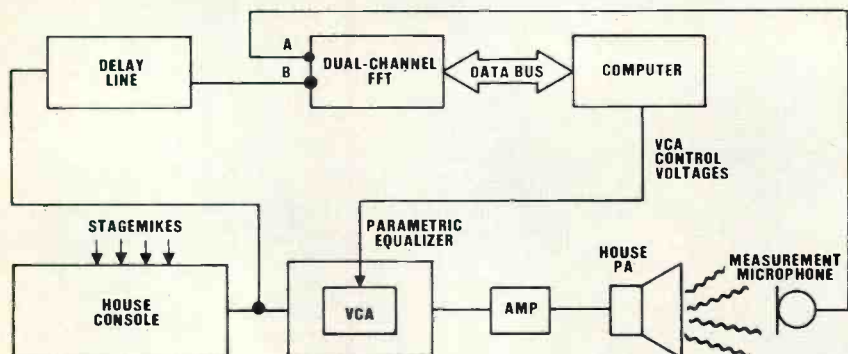
"FFT devices are not able to look at the entire audio frequency spectrum at one time, as are other types of measurement devices such as the TEF [Time/Energy/Frequency] unit," states Dan Healy, the Grateful Dead's sound mixer. "However, for live performances, we have found that 99% of all our EQ 'problems' are going to occur at frequencies below about 1.6 kHz. This makes the FFT a very workable device."

The next step for Dan Healy and Ultra Sound president Don Pearson will be the actual interfacing of the B&K Model 2032 and a personal computer with tuneable precision audio filters. To date, this process has been effected with a single filter in the mid-bass frequency band.

"To do what we want will take some unique and very expensive devices that no one is building yet," Pearson confides. "The cost of the hardware to implement a totally computerized EQ process is prohibitive by most people's standards. We are looking at a variety of options right now."

• **Editorial Note:** As Don Pearson speculates in the above sidebar, computer control of a VCA- or servo-equipped parametric equalizer might represent the next logical step in the development of a self-adjusting concert-sound system. The process essentially would be as follows: First, the degree of correction necessary to compensate for room/system frequency- and phase-response anomalies would be determined from the output of a dual-channel FFT analyzer, using program material during the performance; second, the controlling PC would generate the analog voltages or digital data necessary for an outboard EQ unit to effect the real-time changes in response to, for example, temperature and humidity changes. The one main drawback to the development of such a system, however, is that the cost of component units — not to mention the large number of man-hours necessary to develop the relevant controlling software — might be cost prohibitive for all but the most well-endowed concert-sound company or equipment manufacturer — ML/RT. □□□

Figure 14: Bruel & Kjaer Model 2032 FFT computerized analyzer.

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sets for echo and delay show cues.

A unique noise-gate system has been assembled for the low and high microphone console inputs that handle a Leslie Rotating Loudspeaker on the Hammond B-3 organ. Adjustable gates at the house rack are triggered by the voltage output from the B-3 itself, so that the microphones are gated open only when the instrument is actually being played. Additional dbx limiter and noise gate modules are contained in 900 Series racks for channel-insertion use on bass, guitars, keyboards and vocal mikes.

Ultra Sound Philosophies

"You can start with a basic knowledge of electronics, but it takes getting out into the real world with what you know, and having to work in a variety of situations, to *really* know and understand what it takes to make the entire concert sound system work out here," Pearson stresses. "What we have, when the system is all set up in a hall, is a concert sound laboratory that many equipment manufacturers can only dream about. We are years ahead of some of the major corporations that actually build audio products for the pro-sound user, because we are out here using this system every day, fine-tuning it, improving it. The setup is different every day, too: the power source, the grounding, the hum level from the building, the fly height of the arrays . . . it's all different from night to night."

Future directions for this system may include interfacing an Apple Macintosh computer to the Bruel & Kjaer analyzer, to provide microprocessor control of the system's "tuning" abilities.

"We are looking right now at some precision tuneable filters that are voltage controlled," Pearson confides. "We will be able to enter the parameters within which we want to allow the system to operate.

"As the audience area fills up, and the temperature changes the sound of the system, the computer will have the software to be able to literally make adjustments to the equalization, *without* our hands touching the knobs. That is the direction this system is going."

Pearson explains that much of the forward momentum in terms of technological development in live concert sound must be credited to Grateful Dead engineer Dan Healy. "This band was the first group of musicians in the world to use 16-track recording equipment," he remarks. "Dan has a firm commitment to improving the sound of his show; we have a firm commitment to improving the sound of our systems. We have been working with a group of manufacturers who have that same commitment evident in their product lines. Dan is like the fellow who says, 'What if . . . ?' and we are the guys who take up the challenge."

Subjective Comments

Having recently mixed a concert in this same arena using a well-known traditional sound system, only two months prior to observing the Ultra Sound rig at the Meadowlands venue, I was personally quite intrigued with the prospect of sitting at the console

Figure 15: Meyer Sound Laboratories' CP-10 equalizer.

again. Dan Healy and Don Pearson complied with my wish, and provided this writer with a unique opportunity to compare their event with my own recent memories.

The most notable comment I had upon hearing the performance was that the sound of the room seemed to "disappear." The back-wall, boomy reverberance often heard in such sports facilities was practically non-existent. The absence of muddy reverberation enabled each audience member to more fully focus on the actual musical performance emanating from the stage, rather than feeling detatched from it, as often happens

during rock events for those persons seated in the rear half of the hall.

For the audience members with whom I spoke, the "sound of the show" represented an almost transcendental experience. Just how much of this can be attributed to excellent audio, as compared to the lifestyle of those particular audience participants, is difficult to say.

It would appear, to this writer at least, that a certain synthesis of hardware design and psycho-acoustical effects has been achieved that does make this system sound *different* . . . and perhaps more clear . . . than any I have ever used or listened to before. The application of John Meyer's SIM technique, the meticulous assembly of the Gamble/Crest/Meyer audio-signal path, and the sum total of many years of live sound experience on the part of the system designers and operators, all left me with the impression that I was witnessing a new standard of quality for concert sound. The Grateful Dead performance at Meadowlands Arena would appear to be a benchmark event. ∎∎∎

Author's Note: The mention of specific manufactured audio products in this article is not to be taken as an endorsement. The detailing of such devices and description of their use has been made with reader interest and education in mind — *DS.*

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EXAMINING THE YAMAHA NS-10M "TISSUE PAPER PHENOMENON"

An Analysis of the Industry-Wide Practice of Using a Tissue-Paper Layer to Reduce High-Frequency Output

by Bob Hodas

In the past couple of years, the Yamaha NS-10M loudspeaker has become *the* alternative small control-room monitor. One rarely enters a studio or views a control-room photo in the trades, without seeing a pair of NS-10Ms sitting on top of the console for use as a close-field reference. I have even been in several recording situations where this monitor was the only small reference speaker available, and I was told that my well-known brand of "mini-monitors" were no longer required by most mixing engineers. This being the fact nowadays, there is no reason here to trace the history of why the NS-10M became so popular so quickly. Instead, this article is concerned with the even stranger phenomenon that has followed the rise in popularity of the NS-10M — the use of tissue paper

to cover the tweeters, thus reducing the high-frequency radiation.

As I can best determine, this phenomenon began on the East Coast with an engineer that was having a string of hits, and who spoke of his NS-10M monitoring technique in a number of different interviews. Other engineers then zeroed in on the technique, hoping that it would give them the secret for producing hits. And so the legend grew. This is not merely a practice restricted to the U.S., as one can observe the same occurrence in foreign studios and publications.

The stimulus for writing about this phenomenon finally came when the following interesting story was told to me. It seems that a young engineer walked into the manager's office of a New York studio, and said that he was very unhappy with the monitor-

ing situation. The engineer was using the studio's NS-10Ms, but complained that they had the "wrong" type of tissue paper covering the tweeters.

Well, not wanting to appear ignorant of the latest technology, nor risk being shunned by my peers and *R-e/p* readers, I decided to run comparative tests on different brands and varieties of tissue, paper towels and toilet paper. I went to the grocery store and bought about $30 worth of the above items, borrowed a pair of NS-10Ms from a studio, and proceeded with the tests.

Comparative Test Procedure

To ensure that the best results would be obtained, I decided to use the test facilities at Meyer Sound Labs, Berkeley, CA. Test equipment included a B&K Model 4133 half-inch free-field microphone, B&K Model 2619 pre-amp, Hewlett-Packard Model 3561A analyzer, and an Ashley FET-500 amplifier to power the NS-10M speakers. (For those interested, a complete procedural description of the measurement technique can be found in the October 1984 AES reprint 2150 (I-8), "Equalization Using Voice and Music as the Source," by John Meyer.)

The test included first measuring the speaker in a two-pi position, and then in a position that would simulate their being located on top of a console. Every 20 to 30 minutes during the tests, the reference was recalled to check for fluctuations due to temperature or humidity variations, and was reset if more than a half-dB change appeared. Although only one speaker was used in the tests, we checked the pair to look for inconsistencies. The two speakers measured very similarly; in fact, the NS-10M demonstrates a very linear frequency response with respect to signal level.

Figure 1 shows the two-pi frequency versus amplitude and phase response. The amplitude (top trace) holds pretty close to ±3 dB, and remains quite consistent whether measured at a distance of 12 inches (upper curve) or 24 inches (lower curve). The phase response (bottom trace) is also consistent with respect to level, but notice that between 500 Hz and 4 kHz, there is an excess delay indicating a lack of coherency. This anomaly could be caused by the NS10's crossover circuit, which centers at 2 kHz.

Figure 2 shows the measurement taken at a distance of 24 inches using the final comparison position, simulating a top-of-console location. The curve is almost identical to Figure 1 from 900 Hz and up, but below that point things have changed. While the hole between 180 and 400 Hz has smoothed out nicely, the 500 Hz to 1

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6.5
dB(V)

3
dB
/DIV

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180
DEG

45
DEG
/DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Figure 5.

6.5
dB(V)

3
dB
/DIV

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180
DEG

45
DEG
/DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Figure 6.

6.5
dB(V)

3
dB
/DIV

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180
DEG

45
DEG
/DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Figure 7.

6.5
dB(V)

3
dB
/DIV

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180
DEG

45
DEG
/DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Figure 8.

response when a layer of tissue is placed over the tweeter. The HF roll-off effect begins at around 3 kHz, and drops as much as 3 dB before returning to the reference level at about 11 kHz. Then another dip of about 2.5 dB occurs with a center frequency of about 16 kHz. As we shall see, this curve was fairly typical throughout the test, with amplitude variations being the primary difference. Notice that the phase response exhibits only minor fluctuations, with center frequencies being a bit lower. Tissues that exhibited such response anomalies were two-ply/one-layer versions of Kleenex Pocket Pack Facial Tissue (yellow), Gingham Facial Tissue (white), Nice 'n' Soft Unscented Tissue (white), and Coronet Facial Tissue New Unscented (white).

Figure 5 demonstrates a smoother curve, exhibited by two samples of a single layer of two-ply represented by Kleenex Pocket Pack Facial Tissue (pink and blue). It is interesting to note that the white version of this same tissue —whose response is shown in Figure 6 — exhibited dips 1 to 2 dBs deeper than those detailed above.

Now we'll look at the effect of folding the tissue to form two layers of two-ply covering the tweeter. As one would expect, there is a significant drop in level although, as you will
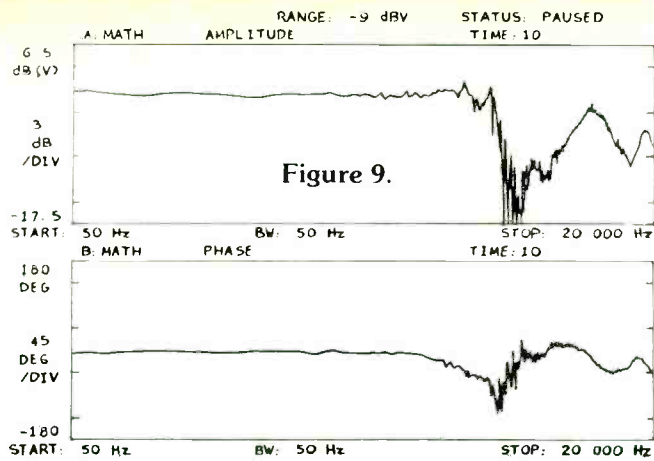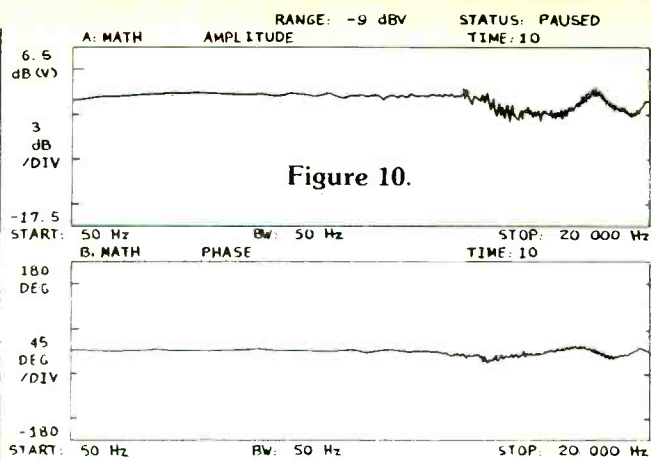
observe, the affected frequencies remain roughly the same. Figure 7 demonstrates this sample group with a 1 dB drop at 11 kHz, and then a more severe high-end rolloff than the one-layer samples. There is also a deeper, wider dip from 3 to 10 kHz. Samples in this group were Kleenex Pocket Pack Facial Tissue (blue, yellow, and pink), and Coronet Facial Tissue New Unscented (white).

Figure 8 demonstrates a more extreme effect. While the 11 kHz region is hardly affected, the two dips have become more severe. Gingham Facial Tissue (white) and Nice 'n' Soft Facial Tissue Unscented (white) fit into this sample. The real surprise in this sample was that two of the one-layer brands exhibited this same curve: Kleenex Facial Tissue (white), and Kleenex Softique Lightly Scented (yellow).

Figure 9 shows the most extreme effect in the two-layer tests. As can be seen, the rolloff is extremely non-linear, and the dips are quite deep with these samples. Represented here are Kleenex Softique Lightly Scented (yellow), and Kleenex Facial Tissue (white). The effect is not surprising in reference to these two tissue's performances as single layers in Figure 8. Somewhat mellower — but still extreme — were Kleenex Boutique Facial Tissue (blue) and Kleenex Pocket Pack (white).

From these test results I found it difficult to draw generalized conclusions based on brand, model or color, and so proceeded to measure the toilet paper samples to see what that might uncover. The T.P. samples grouped themselves into four categories, and we will look at them in order of the degree of effect they had on the reference.

Figure 10 represents the first group, and one will notice immediately that the effect is less than that produced by any of the tissue groups. I would have to attribute the difference to the fact that these were all one-ply paper, while tissue is always two-ply.

6.5 dB (V)

3 dB /DIV

Figure 11.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

6.5 dB (V)

3 dB /DIV

Figure 12.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

6.5 dB (V)

3 dB /DIV

Figure 13.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

6.5 dB (V)

3 dB /DIV

Figure 14.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

6.5 dB (V)

3 dB /DIV

Figure 15.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

6.5 dB (V)

3 dB /DIV

Figure 16.

-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
B: MATH   PHASE   TIME: 10

180 DEG

45 DEG /DIV

-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Included in this group were Charmin New Stronger Than Ever (yellow and blue), and Family Scott (white).

Figure 11 demonstrates that even two-ply T.P. does not approach the 3 to 10 kHz drop caused by tissue, yet is just as effective at the dip above 11 kHz. T.P. in this group was Northern Quilted (yellow), MD Unscented (white), and —a big shocker — Scottissue one-ply (white). As some of you may already know, this latter brand is a hefty one-ply, and the result most likely due to its beefier construction.

Figures 12 and 13 show the effect of using the T.P. in a two-layer configuration, and these samples more closely resemble curves that one would expect

from of tissue paper; they exhibit the wide 3-dB holes from 3 to 10 kHz, and the deep dip after 11 kHz. Figure 12 is represented by one-ply papers: Scottissue (white), Family Scott (patterned), and Charmin New Stronger Than Ever (blue and yellow). Figure 13 is represented by two-ply papers, illustrating only Northern (yellow), and MD Unscented (white).

I almost didn't measure the effect of paper towels, because no one has ever mentioned using them. But, for the sake of science, I decided to explore new, uncharted territory. Every tested paper towel exhibited a different curve, showing no consistency for drawing conclusions. One very inter-

esting thing was found, however: Brawney two-ply in a two-layer configuration exhibited the smoothest rolloff of any test sample, as can be seen from Figure 14.

So what conclusions can we draw at this point? We can rule out color as a factor in the measurements, but cannot definitely state that two layers will cause much deeper frequency-response dips than the same tissue or paper in a single layer (Figure 8). We can see that, at times, two layers of T.P. can simulate tissue curves, but cannot draw positive conclusions about the effect of ply number (Figure 11).

At this point I was generally

Figure 17.

RANGE: -9 dBV   STATUS: PAUSED
A: MATH   AMPLITUDE   TIME: 10
6.5 dB(V)
3 dB/DIV
-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

B: MATH   PHASE   TIME: 10
180 DEG
45 DEG/DIV
-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

Figure 18.

RANGE: -9 dBV   STATUS: PAUSED
A: MATH   AMPLITUDE   TIME: 10
6.5 dB(V)
3 dB/DIV
-17.5
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz

B: MATH   PHASE   TIME: 10
180 DEG
45 DEG/DIV
-180
START: 50 Hz   BW: 50 Hz   STOP: 20 000 Hz
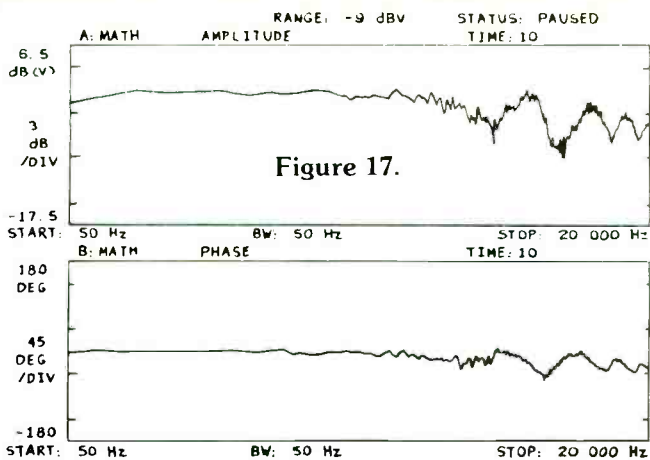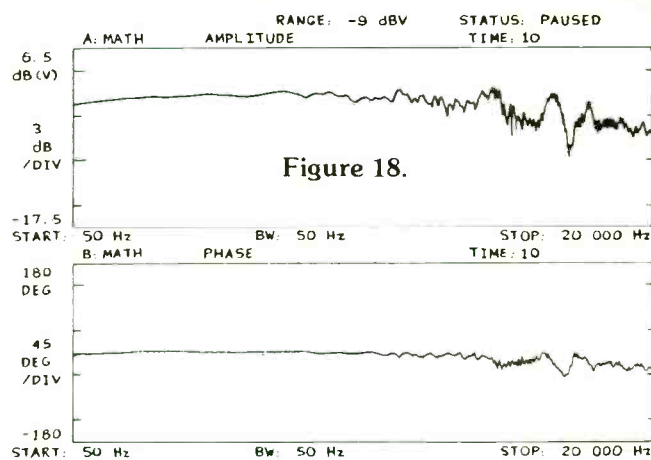
unhappy at not being able to draw more precise patterns from this experiment, and so I took another approach. Thinking that I may get a more accurate measurement of high-frequency absorption, I placed the tissue over the microphone. Imagine my surprise when the chart came out looking exactly like Figure 3, and exhibited absolutely no deviation from the reference. This discovery led to the idea of hanging a sheet of two-ply Kleenex in front of the tweeter at distances of 0.5, 1, 1.5 and 2 inches; the results of these tests are depicted in Figures 15 thru 18, respectively. Interesting, don't you think?

So now we have something concrete: Kleenex did *not* absorb any high frequencies when placed over the microphone capsule, while Figures 15 thru 18 indicate that the HF attenuation affect of tissue is position dependent. As a result, it would appear that the tissue creates the one thing that studio designers around the world try to avoid, fearing it most in their control rooms: *Comb Filtering*. The paper filters are *not* absorbing the high frequencies, but simply reflecting them back into the tweeter, thereby causing cancellation through comb filtering.

Since nobody would put up with such aberrant behavior in their main control-room monitors, why should we go this route in a close-field speaker? The result is a very crude and, I would say, undesirable method for dealing with the problem of a speaker sounding a bit too bright. And, as one can see from the charts included in this article, the use of tissue layers is certainly a very uncontrollable and nonlinear method of doing the job. If a little less high-end is desired, how about a simple electronic filter that could be measured and controlled in a more reliable manner than the "Tissue Fix?" ∎∎∎

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RECORDING AND PRODUCTION TECHNIQUES FOR THE

BOSTON SYMPHONY ORCHESTRA

by Paul D. Lehrman
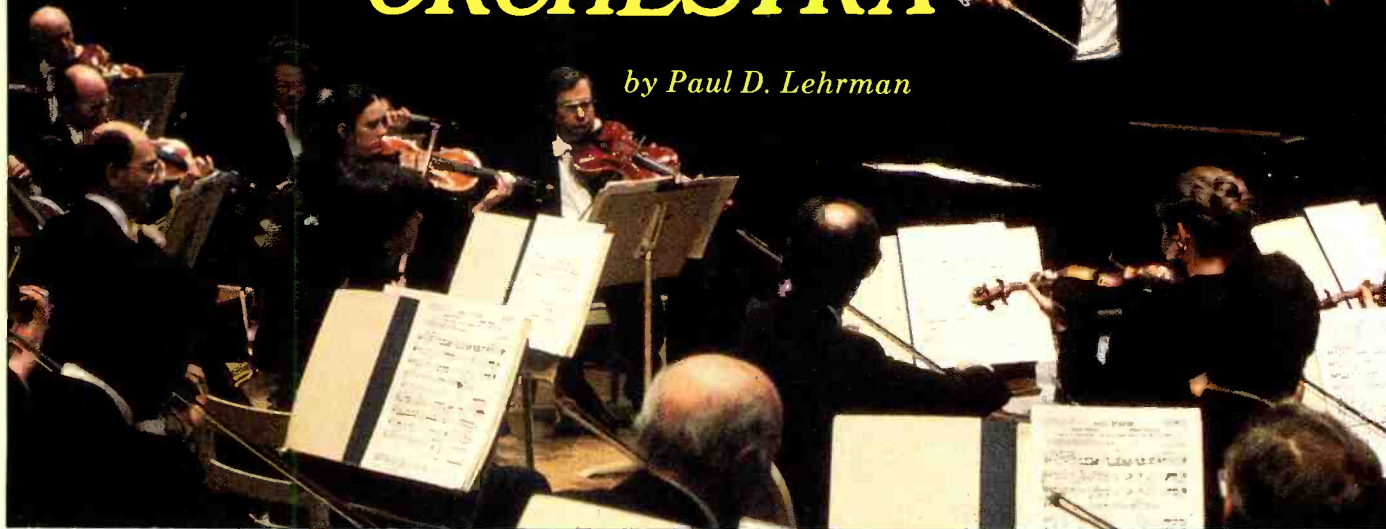
How do you record the Boston Symphony Orchestra? "Don't ask me," says Richard Kaye, who has handled that duty for the BSO more than anyone else. He names a couple of well-known local audio critics. "Ask them," he laughs. "They're always telling me how to do it!"

Kaye is the producer for the Boston Symphony Transcription Trust, the Organization responsible for dozens of live broadcasts and recordings for radio that the BSO does every year. As such, Kaye has his work cut out for him when trying to please this large and diverse audience. In Boston, the Science of audio and the Art of classical music are uniquely melded together; there is probably more classical radio programming *per capita* than anywhere else in the world, thanks to an established, successful commercial radio station (WCRB, where Kaye is president), and a plethora of public radio stations, many of whom — most notably WGBH-FM and WBUR — broadcast a substantial amount of classical programming. The surrounding area is populated by numerous pro-audio and high-end, hi-fi manufacturers, plus performance groups ranging from a world-class symphony to dozens of chamber and new-music ensembles. And the city boasts one of the best-

sounding performance spaces in the world: Symphony Hall. In the middle of all this are troupes of audio writers and music critics.

And those writers and critics love to make their own recordings. It's not unusual, says WGBH engineer Bill Busiek, for there to be five or six different recording setups taping an orchestral concert in Symphony Hall — although it doesn't happen with BSO concerts. "We've had someone from *Stereo Review* using three spaced Shure SM-80s," he says. "Someone else using ORTF, and someone else with a Blumlein pair — all recording the Boston Philharmonic [an ensemble that performs a few times each year]. All they have to do is get permission from the orchestra, and pay the hall fee."

With all of that competition, producers of the BSO's records and broadcasts have to be on their toes, and not be afraid to experiment. Fortunately, they have plenty of opportunity to do so. Twice weekly, the BSO gives live and live-on-tape radio broadcasts from Symphony Hall, and three a week from its summer home at Tanglewood in western Massachusetts, as well as doing records for several domestic and foreign labels. The Boston Pops, which is essentially the BSO without its first-chair players, does live and taped radio and TV broadcasts dur-

ing its summer and Christmas seasons, and plenty of records, thanks in large part to the popularity of its two most recent conductors: composer John Williams and the late Arthur Fiedler.

And both orchestras share an illustrious heritage: last year the Boston Symphony celebrated its Centennial, a landmark that was passed by the Boston Pops during its 1981/82 season.

Although there is some overlap, each of these ways of relaying the orchestra's sound to the public involves different personnel, different equipment and program chains, plus different production philosophies and styles.

Live Radio Production

Live broadcasts from Symphony Hall are a joint effort between WCRB and WGBH, whose services are contracted by the Boston Symphony Transcription Trust. Bill Busiek operates a 24-input Neve console that was installed in the small radio booth overlooking the stage about seven years ago. Richard Kaye sits next to Busiek, following the music score and the voice-over script, which is read in ringing tones by announcer William Pierce located in an adjacent booth.

During the week preceding each broadcast, WCRB engineer Marian

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AND THE BEAT GOES ON

R-e/p 64 ☐ February 1986

RECORDING THE BSO

"Tuna" Howell attends several concerts and rehearsals, taking levels and noting in Kaye's score if they will have to be adjusted. "The faders stay virtually untouched," confides Busiek. "When we do move them, it's only a few dB. Sometimes the timpani or the horns get a little too well reinforced by the hall, and we have to compensate. We'll do the changes very gradually, way beforehand, like at a rest. That way a whole piece or movement doesn't have to be down in level, but at the same time the audience doesn't hear a crescendo or decrescendo."

Five microphones are semi-permanently installed in the ceiling of the hall, and wired directly to the radio booth. Although the lines have been in place for years, the mikes themselves are new: three Bruel & Kjaer Model 4007 small-capsule omnis are mounted in a line over the stage apron, while two more B&K Model 4006 large-capsule omnis are placed further out into the audience, where they can pick up some of the bass developed by the hall. Some of the mikes are owned by the broadcasters, while others are on-loan from the manufacturer.

The current season is the second in which the three-mike format has been used. Previously, there were two main mikes, usually Neumann KM83 omnis. "There are incremental advantages to using three mikes," Busiek explains. "You have absolute assurance of mono compatibility, and imaging is easier — there are always angle and distance factors to take into consideration, and using three mikes limits them without compromising the sound. There's something solid about it, and you get a better perspective on the orchestra. You take the risk out, and the only disadvantage is that you have to worry about an extra mike. We're using the front mikes only now. We put the orchestra in intensity balance, but it means we lose a little of what the hall sounds like.

"X-Y miking is wonderful, and it'll always sound stereo, but it's not so easy to get the correct balance. Sometimes it's *too* stereo, and there's a hole in the middle, and sometimes it's too mono. Also, since you're using the sides and lobes of the [cardioid-pattern] mikes that way, they have to be perfect [in terms of even on-and-off-axis frequency response], and it's an imperfect world."

Busiek explains that the center mike is mixed a little lower in level

than the outside ones — "If it was the same level, the middle would be too heavy" — but he is reluctant to state precise details on the levels. "We're not in a position to say 'This is it' about our miking," he says, "and we aren't discouraged from experimenting. But we *are* limited by what is practical. There are 'political' issues, like who has access to the mikes" — they hang through trap doors in the ceiling from lighting beams and other supports —"and how often you can get up and change them."

"The hall opened in 1900," notes Kaye, "and the plaster ceiling is literally held up by horsehair, which means we have to be *very* careful about going up there and drilling holes."

Monitoring in the booth is handled by ADS Model 900 speakers, driven with a BGW Model 500 power amp. Two old-reliable AR tuners are located in the booth, permanently tuned to WCRB and WGBH, but they aren't used too much. "I'd like to monitor off the air," says Busiek, "but it would give me ulcers. We just use them for cueing."

From the radio booth, telephone company lines go to WGBH's studio in Allston, 2½ airline miles away, and WCRB's studio in Waltham, about nine miles distant. WCRB's lines are processed with Dolby A-type noise reduction. "WGBH used to Dolby [its lines] too," says Busiek; "encoding here and not decoding until the signal got to the transmitter. But there was too much room for error, especially when we would inject intermission

features at the studio, and we discovered our lines were good enough that we didn't need it.

"If the lines are in trouble, we can switch the [units] in, but generally it's more trouble than it's worth. They can exaggerate low-end anomalies, and they have to be calibrated more accurately than the world thinks; listeners are very quick to pick up mistracking."

The WGBH signal is relayed to the station's transmitter atop Great Blue Hill on Boston's southern city limit (about 10 miles from the studio) via two separate Moseley microwave STLs, one for each audio channel. But when conditions are right, WGBH uses a totally different transmission system: digital encoding with a dbx Model 700 convertor.

"dbx's method [Companded Predictive Delta Modulation] makes a lot more sense to me than PCM," says Busiek, although, as he admits, "I can't really hear the difference." What has made the Model 700 the station's choice is the fact that dbx is a local company, and its designers and staff have been particularly helpful in setting up and maintaining the system. "And it's true balanced-in and -out," adds Busiek, "so it doesn't unbalance the rest of the system."

WBGH does not decode the digitized signal (in composite-video form) until it reaches the transmitter, bouncing it from a dish on top of Symphony Hall, to the top of the nearby Prudential Tower, to the Allston Studios, and finally out to the transmitter. "We do it all 'blind'," says Busiek. "We don't

WGBH engineer Bill Busiek (left) and WCRN producer Richard L. Kaye during the recording of a BSO concert in the radio booth at Symphony Hall. Seen in the background is announcer Wiiliam Pierce.

decode it at the station; we just monitor it as a video signal, and listen off the air. All of the switching is done by remote-control over the microwave STLs."

Busiek doesn't get to use the system as often as he would like, however, the reason being that in a city with as crowded a spectrum as Boston, it's not always possible to get a clean video path. "We share the Prudential with a lot of other TV services, including WGBH-TV," he explains. " All it takes is for someone to be doing unauthorized testing on one of the path frequencies, and it blows us out. The unit sees too much error, and just mutes. We do 'border' broadcasts of two minutes of crowd noise before the concert starts. If we hear any drop-outs, we switch right back to the 'phone lines."

"We're not discouraged at all," the engineer continues. "We're working on overcoming the legal restrictions, and trying to get a different route that will bypass the Prudential and get us a direct shot to Allston or Great Blue Hill. It just takes money."

WCRB is the only commercial broadcaster that carries the live broadcasts — the others are all public stations. Five stations in Maine and one in Connecticut pick up the signal from microwave landlines, and one station on the border with New York state simply rebroadcasts WGBH's air signal — a practice that Bill Busiek, understandably, disapproves of.

Summer concerts at Tanglewood are handled much the same way. Because the ceiling of the performance "shed" is a triangular grid, there is much more flexibility in hanging mikes. The B&K models come along, while the mix is handled by a brand-new Studer Model 264 12-in/four-out console. The signal for the live broadcasts is passed to a microwave link located on the grounds, which is owned by WGBH. There are two short microwave relays, and then the signal is processed digitally with the dbx Model 700 and put on a video channel originating on top of Mt. Tom, for the long final hop into Boston. "It has to go over the Quabbin Reservoir," explains Busiek, "where there is often fog and thunderstorms, both of which can severely attenuate a microwave signal."

Pre-recorded Concert Performances

At the same time the concerts are going out over the air, Tuna Howell is putting them to tape. Although, during the regular season, only the Friday-afternoon and Saturday-evening concerts are broadcast live (and the latter carried only by the two Boston stations), the orchestra usually performs the same program two or three more times earlier in the week. Almost all of these concerts are recorded. Three tape formats are used simultaneously: 15 ips half-track Dolby A and non-Dolby on a pair of venerable Ampex AG-440 decks; and a digital backup copy made with a Sony PCM-701 processor and SL-2000 Betamax VCR.

Decisions on how to edit the tapes are worked out by Kaye and the concert's conductor, whether it's BSO music director Seiji Ozawa, or a guest. "We don't do very much," says Kaye; "typically about two edits on a program."

During the tapings, Kaye will mark potential problems in his score and script. "For really bad mistakes, especially if the crowd noise stays the same under the voice-over, there's no problem cutting something," says Howell. "We don't cut inside movements, but we'll sometimes use one movement of a piece from one concert, and the other movements from another. If there's a minor flub that would take 30 minutes to fix, we won't bother."

The edits are performed on the original masters, to keep generation noise at a minimum. If a gross editing error occurs, a new master can easily be generated from the digital-701 copy.

Tapes are duplicated at WCRB on Asco Series 2400 16:1 duplicators, and go out to some 80 stations on a weekly schedule, generally about two months after the actual concert. Of these 80 stations, less than a half-dozen receive Dolby A-encoded versions. "It means that all of the edit instructions have to be performed twice," says Howell.

The two Ampex tape decks generally do not make the trek out to Tanglewood for the summer — instead the concerts will be recorded on an Otari MX-5050 that spends the winter in the radio booth at Symphony Hall as a backup machine. Kaye takes the Sony PCM-701 digital system along too, and he will often bring in a two-track Technics RS-1500 analog deck. Another machine, often one of the Ampex AG440s, will be brought in for editing. The Tanglewood tapes become part of the program rotation, and are played over the subscribing stations later in the season.

Television Audio Production

Once upon a time, *Evening at Symphony*, a series of live and taped BSO performances, was a staple of PBS television stations around the country. The program was known for its technical and stylistic innovation, both in the audio and video realms — it was reportedly the first regularly-scheduled TV/FM simulcast series, each local radio station being fed with signal from a four-track audio tape striped with sync tone or SMPTE timecode — and brought the sight and sound of symphonic music into the

Engineer Bill Busiek with a video transmitter unit at Symphony Hall used for relaying digital audio encoded with a dbx Model 700 CDPM processor back to the WGBH studio.

Paul D. Lehrman
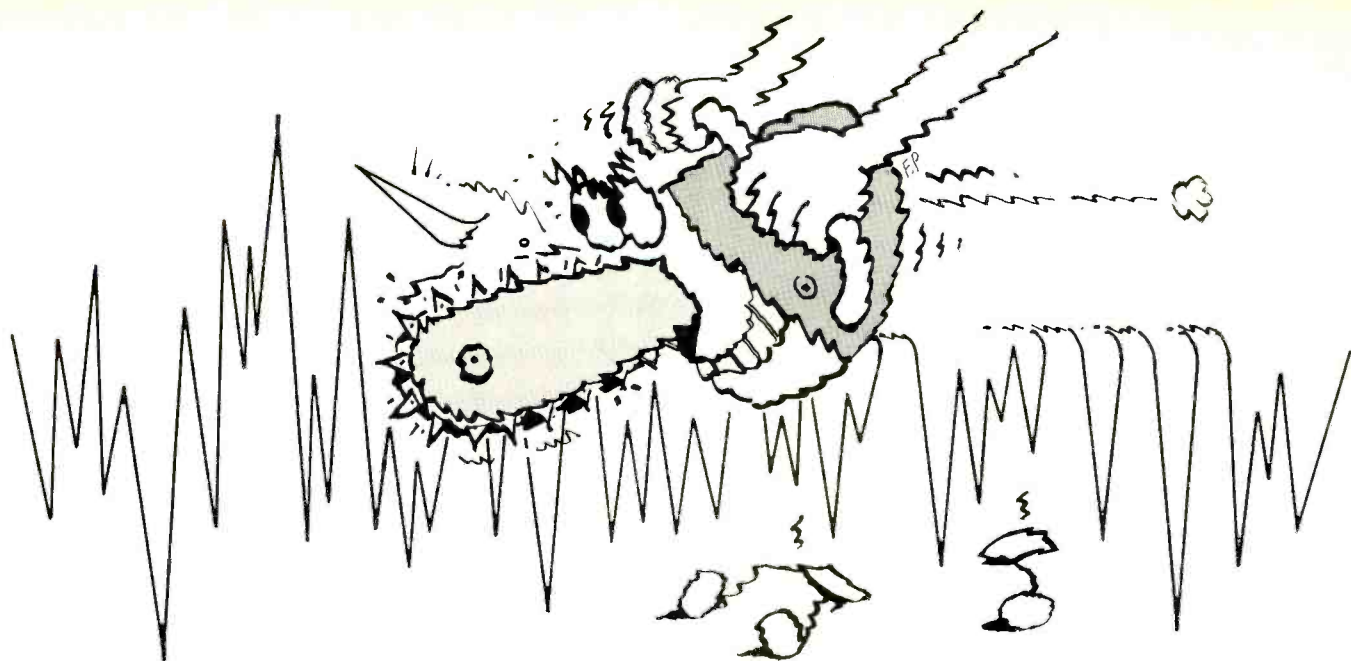
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homes of millions for the first time. As the decade turned, funding cutbacks reduced the series to the status of occasional "specials," and eventually it was discontinued completely.

Today, the only regular television programs from Symphony Hall are the Boston Pops concerts, about a half-dozen of which are taped each year during the Pops' summer season. In addition, there is usually one live broadcast each season, such as last summer's Fourth of July gala from the Charles River Esplanada.

Evening at Symphony, under the direction of the late Jordan Whitelaw, pioneered a number of audio and video techniques, including quadrophonic sound, multitrack recording, and careful "orchestration" of camera angles and video switching to enhance greatly the viewing experience. "The concerts were originally taped on 16-track Ampexes," explains Steve Colby, who is the supervisor of WGBH's mobile audio unit [see: *Audio Production for Broadcast,* Spring 1983 issue for further details], and current audio engineer for the Pops TV broadcasts. "There would be an overall stereo pickup," he explains, "plus spot mikes, which would vary depending on the piece. There would be tracks for vertical drive and for timecode.

"The tape would be mixed down to two-track on the same piece of [16-track] tape. Since we mixed off the playback head, the audio would be placed about five frames off [sync with the individual tracks and master timecode]. It was a matter of convenience, in that we just had to worry about one piece of tape. I understand that before there was timecode, they would use only the sync pulse to lock the audio and video. At air time, an engineer would sit by the tape decks with a pair of headphones, listening to countdowns — audio tape in one ear and the video in the other. He would have 60 seconds to match them up."

Today, Pops concerts are recorded on 24-track Otari MTR-90s in the WGBH mobile. SMPTE timecode has made the engineers' lives more tolerable, but otherwise the techniques have changed very little. The concerts have been mixed in both mono and stereo for the past four years, although until recently they were broadcast almost exclusively in mono — the shows were distributed to the stations on videotape, and FM/TV simulcasts were extremely rare.

Since last year, however, the pro-

grams have been distributed entirely on the PBS satellite system, and every city receives stereo audio, whether they use it or not. Some markets, such as German television, have used the stereo mixes for some time, and, says Colby, "We wanted the archives in stereo."

The mixing process is somewhat more active than that used for Symphony concerts. "Pops shows need more control, more presence," says Pops TV-audio producer John McClure. "When you show an instrument on a close-up, it shouldn't stick out grotesquely. But you have to hear it, and that means multitrack."

"We want to achieve a partnership between audio and video," Colby says. "It should be a good stereo mix that will stand on its own, so that even if you're not watching, the audio should make sense, without something jumping out at you. We never exaggerate something to match the picture if it's not going to make sense in the mix. It's a kind of 'soft cueing' — we help the viewer make an association with the picture, but the faders never move more than 3 or 4 dB."

A typical track sheet will contain a main audio pair and two tracks each for strings, woodwinds, brass, and the audience. Harp, percussion, and trap set each get one track, while a guest performer or group will get up to four tracks. Vertical video-sync drive and SMPTE timecode, fed from the video truck, go down on tracks #23 and #24, respectively.

Some of these stereo pairs actually are mixed from four or five microphones. Most of the mikes are hung

from the hall ceiling, with floor mikes being used on harp, piano, drum set, brass, and the guests. "We tend to use the mikes that 'GBH has available," says McClure, "which are [Neumann] KM83 omnis, KM84 cardioids, and AKG 451s and stereo models."

"I'm dying to use those new B&Ks," adds Colby.

Permanently-installed cables run from the stage ceiling (the "rope loft") to an outlet box mounted on an outside wall of the hall at street level, where the WGBH mobile parks. Some of the cables have individual splitters on them so that they can feed both the radio booth and the mobile unit. Signals from the floor mikes are routed through a snake that runs underneath the stage to a splitter box, and then to the house PA mixer located in the first balcony, the monitor mixer directly in front of the stage, the radio booth, and the truck.

The multitrack tape is mixed in the mobile unit after the concert, sometimes within a day or two, sometimes not for a couple of weeks. Tracks #21 and #22 are reserved for the two-track mix, again to simplify tape-handling. During the actual concert, these tracks are used for a live mix. "We can always refer back to it," says Colby, "and sometimes, if it's really hot, we'll end up using it instead of wiping it and doing another mix." Often the guest artists, from Noel Paul Stookey of Peter, Paul and Mary, to members of the Manhattan Transfer, will be involved in the actual mixing process.

Although there are some shared mikes, radio and TV production are almost always handled completely

Music director Seiji Ozawa conducting the Boston Symphony Orchestra

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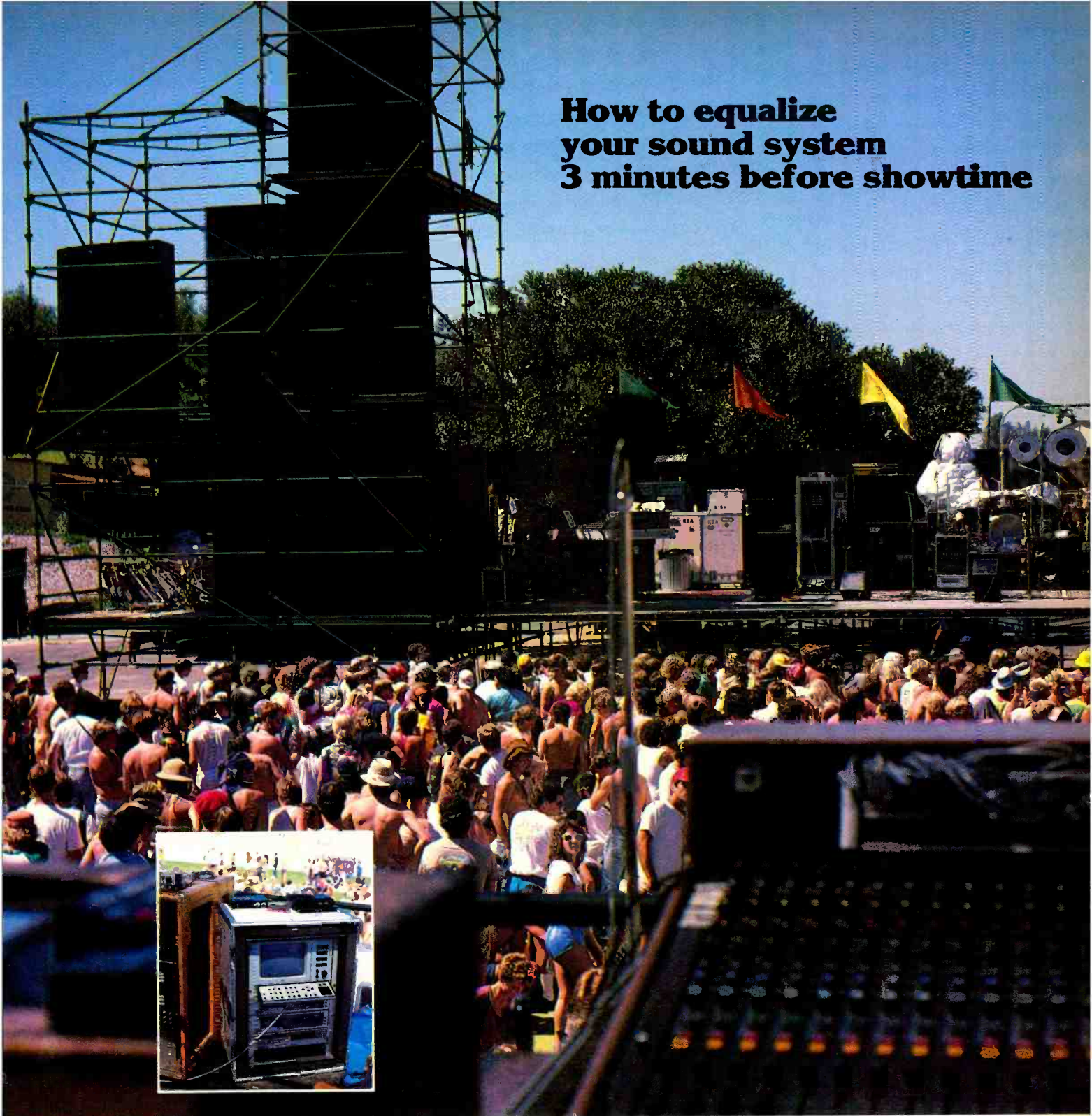
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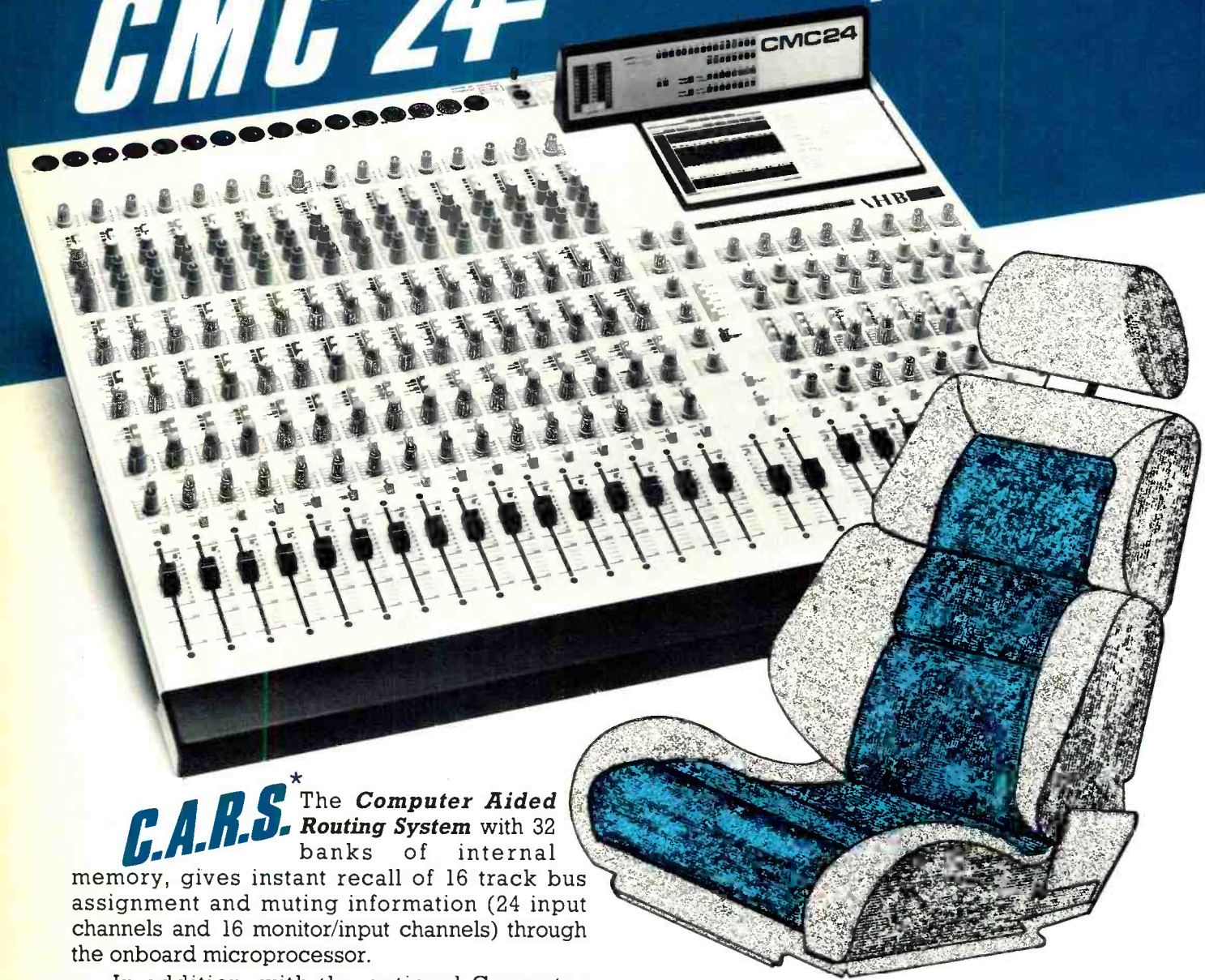
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RECORDING THE BSO

separately. "When we did the Man-hattan Transfer, they were so involved in the TV mix that they insisted that it be used for radio too," says Colby, "but that's rare."

Generally speaking, the guidelines for assembling a Pops TV show are less strict than those for the Sym-phony's radio broadcasts. "Usually we do one concert per show," offers Colby, "but sometimes not."

There may also be a certain amount of editing within the body of a piece to overcome technical or performance problems, something that is almost never done with Symphony broad-casts. "Like any editing problem," Colby says, "We hope the segments fit together. We're lucky in that the hall maintains its acoustic characteristics every night all through the Pops sea-son, unless it happens to get really cold or humid. One night we noticed that the sound we were getting was particularly terrific, and we figured out it was because it was pouring rain outside.

"Because we're close-miking the soloists, it eliminates the second-by-second hall effects. We'll sometimes add a little reverb to make it work. We're not proud; we don't think it's breaching the artist's integrity if we use a technical aid to make it sound more natural. If we have to edit — like when a waitress pops a champagne cork during a vocal solos — we can strip the hall tracks and replace them with a Lexicon 224 [digital reverb] popping the two in and out. I hear that program number #3 on the Lexicon was the designer's idea of what Sym-phony Hall sounds like and, eight times out of 10, it's a perfect match."

McClure and video producer Bill Cosell confer on how the shows will be edited, and McClure and Colby then mix the concerts while the final video edit is assembled. "We'll give the video people a complete mix, and then they punch the numbers into the computer and do the wedding," says Colby. [For more on WGBH's time-code editing facilities, see *Audio Pro-duction for Broadcast*. Spring 1982 issue.]

Classical Recording Sessions

The BSO does not make as many records as it used to, which is in keep-ing with the general state of the classical-record industry in the U.S. In the 12 years that Ozawa has been at the helm, the orchestra has recorded for Philips, RCA, Angel, and Hyper-ion, while Deutsche Grammophon

For Telarc Records first BSO session in January 1980, piano soloist Rudolf Serkin is seen here playing Beethoven's Fifth Piano Concerto, with Seiji Ozawa conducting.

held an exclusive contract for a few years in the Seventies. Today, the label most actively recording the group is Telarc. There have also been dates for CBS and New World records and these, interestingly enough, have

been engineered by Telarc's president and chief engineer, Jack Renner.

Telarc began its association with the BSO in 1980 when it began a com-plete cycle of Beethoven Piano Con-certos with the orchestra and soloist

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Rudolph Serkin, which the label finished last year. Since the beginning, Telarc has used the Soundstream digital recording system, and minimal miking.

"I'm very much a believer in the idea that less is better," states Renner. "The minute you start putting up more mikes, you take control away from the conductor — you intrude between the orchestra and what the conductor is trying to achieve."

Renner generally uses three Scheops SKM55U mikes switched to omni pickup (although lately he's been experimenting with WGBH's B&Ks), placed up on stands about five rows into the audience. "Musical instruments need a certain amount of 'space' to develop their characteristic sound," he explains. "If you mike too close, you won't let the harmonic characteristics of the instrument or the ensemble develop, which will let people recognize it." Renner knows whereof he speaks: both he and Telarc producer Robert Woods are former performing musicians, conductors, and teachers.

"We use omnis for their flat frequency response, especially at the low-end, and their better distortion figures," Renner says. "They let us capture a good balance between the direct sound and the hall acoustics. A symphony orchestra is a wide beast. To get a good representation of it, you have to spread mikes; if you're only using two you leave a hole in the middle."

For the latest Beethoven recordings, which have been of the earlier concertos, Renner has been able to get away with using only two mikes, because the score calls for a reduced orchestra. "We also put the bass drum in the middle to control the vertical information when we're cutting an LP, and to keep the cutter head from jumping out of the groove. Of course, that's becoming less important as we leave the age of the LP."

That last statement may sound a little premature but, as far as Telarc is concerned, vinyl albums are pretty much on their way out — Renner says that his Compact Disc sales outnumber LPs by nine-to-one, and the entire Telarc Catalog was made available on CD last fall. Renner also says that when it comes to deciding program lengths, his thinking is geared more towards CDs. "We're putting a minimum of 50 to 55 minutes on each disc." But the label isn't quite ready to leave LPs behind completely: Telarc is taking advantage of Teldec's direct metal mastering (DMM) process to increase vinyl playing lengths as well. "DMM has been our salvation," Renner says. "So far, we've been able to duplicate everything on vinyl, and some of our sides have 38 minutes of music!"

For this kind of session, the recording environment is all-important. "It puts a burden on the engineer to find a good hall," the engineer states. "An excessively reverberant hall, like a stone church, won't work right — the sound gets muddy two feet from the stage. Symphony Hall is right on the edge of being too reverberant."

For its early recordings, Telarc would lay huge carpets and sheets of burlap over the front of the balconies to dampen the sound of the empty hall, but Renner says that's no longer necessary. "There's enough detail and presence coming from the stage so that we don't have to do that, and we can work slightly closer. We also take out the risers that the orchestra usually sits on, which tend to be bass reinforcers."

Although Soundstream is no longer in the digital-hardware business, Renner still uses the company's PCM recording system. He considers its sonic qualities, especially in the crucial area of A-to-D conversion, to be superior to all of the others. He's not worried about maintenance — "I know where the bodies are who can keep the [hardware] running, and where to get spare parts."

He no longer edits with the system, however, primarily because the only Soundstream editing facilities are in the U.S. are located in New York and Salt Lake City, a long haul from Telarc's base in Cleveland. Instead, signals from the Soundstream are transcodes in real-time to Sony PCM-1610 format through a Studer Sampling Frequency Convertor (which Renner says doesn't affect the sound) and recorded on U-Matic videotape. Back in Cleveland, the tapes are edited on a Sony DAE-1100 system. "It allows us to edit here," says Renner. "And besides, the 1610 is the only acceptable format for CD mastering."

For the Serkin sessions, the orchestra's "Green Room" at the side of the stage serves as a control room, while on the other dates Renner uses a recording booth built (and later abandoned) by Deutsche Grammophon in the basement underneath the aud-

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ience. "There are permanent mike lines from the stage to the basement." says Renner. "They have Tuchel connectors on them, so we get John Newton [a local engineer whom we will talk about later in this article] to bring in the adaptors he has." From the beginning, Telarc has used a Neotek Series Two console and ADS monitors, although on some sessions — especially those done in Europe — B&W Model 801 speakers will be used.

Backup recordings used to be handled by Studer analog recorders, but today they are done digitally, using a Nakamichi PCM processor and half-inch videotape. "We've never had to use the backups," says Renner, "But I wouldn't hesitate if we had to. Personally, I think they sound better than the 1610, anyway."

General editing decisions are primarily in the hands of producer Robert Woods. "During playback, Bob and the conductor will talk about the takes," explains Renner, "but they don't sit down and go over them note by note. But, if the conductor is con-

cerned about anything, we'll know about it before we leave. We try for long takes, and get at least two complete takes of each movement. Then we do inserts as needed. In the final product, it comes across better musically to keep it as long as possible between edits.

"The broad decisions are done right at the sessions, and then Elaine Martone, our head of production, who also has a musical background, goes through Bob's suggestions. If she has any problems, she'll look to the other takes. She'll give Bob a rough edit to listen to and then it goes out to the conductor for him to hear."

The procedures don't change much when Renner is recording for labels other than his own, although he is working with different producers. "The editing is done in New York, and sometimes the Studer SFC isn't available, so we'll do everything with the Soundstream system."

Recording the Boston Pops, with its far more diverse repertoire, can be a totally different experience from the way the BSO is handled, or it might be strikingly similar. Pops releases under John Williams have ranged from Rossini overtures to the theme from

E.T., and each type of music requires a different approach.

The Pops are under exclusive contract to Philips Records, and TV-Audio producer John McClure also serves as record producer. McClure has plenty of experience as a classical producer, having once been department director for CBS Masterworks, where his duties included producing several BSO recordings. Like the BSO, Pops recordings are done digitally using the Soundstream system but, in this case, the equipment is available locally from Soundmirror, a company owned to the aforementioned John Newton, and the system is usually used right on down the recording chain.

"The recording philosophy varies with the subject matter," McClure says. "If it's a straight symphonic session, we'll do it like Renner. But we can't always afford to take a rigid, 'purist' attitude, although it would be nice if we could — to sit back and let the conductor and the orchestra do all the work. If we've got a record with a swinging brass section, for example, it's another whole ball game.

"The idea is to use the minimum necessary to achieve your objective,

but not to be afraid to use more [equipment] if you need it. A muted trumpet solo or a hi-hat and snare track doesn't come through well with an overall pickup, and [the sound of] trap drums rolls around the hall like they're in a cavern."

The solution is accent mikes, using up to a dozen Schoeps Colette (CMC Series) omnis for the purpose, and running them through Newton's Studer Model 169 console. Rather than depending on multitrack and post mixing, however, Pops records are taped direct to two-track or, on some occasions, to four-track. Examples of the latter, where a little post-session flexibility is desirable, are a record with vocal soloist Jessye Norman, a collection of patriotic numbers with chorus and soloists, and a disk due for release next Fall featuring the orchestra in conjunction with a big band.

Backup tapes are made on a Sony PCM-F1. When the session is done, says McClure, "John Williams and I sit down with an F1 cassette and do an edit list. We send the list and the Soundstream tapes out to Salt Lake, where Tom McCluskey does the edits, and then sends the tape back to Bos-

ton. John Newton and I then do the sequencing and, if it's a four-track tape, the mix. We make analog cassettes for Williams and any guest soloists for their approval, and then send the whole thing off to Philips in Holland for mastering.

"There's a Soundstream system over there, so they can use the tapes for disk mastering, but obviously if they're going to do a CD, they have to convert to the 1610 format through one of the Studer [SFC] boxes. Sometimes, if they can't get the Studer unit, we'll do the conversion here using Renner's. On the last record we did, we mixed the four-track directly down to the 1610, but that's the only time we've ever done that.

With Soundstream's future in doubt, McClure is unsure of his own plans. "There's still the system in New York, and there's talk that some investors from Canada may put some money into it, reviving and maybe even upgrading it. I hope so — it's a wonderful editing system, once you get everything loaded in."

And even the audio critics will agree that with the right equipment in the right hands, the best BSO and Pops recordings sound awfully good.■■■

From the January 1980 Telarc session with the BSO: Seiji Ozawa and Robert Woods (seated); Jack Renner, Telarc president and chief engineer, and Dr. Thomas Stockham, former president of Soundstream, Inc., whose digital PCM recording system was used on the session.

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EMERGENT TECHNOLOGY

Thanks to the unprecedented worldwide acceptance of the digital audio Compact Disc, a startling new technology is emerging that uses CDs for data and program storage. The CD-ROM (Compact Disc Read-Only Memory) will truly advance the technology of computerization in terms that can only be described as a "quantum leap" forward.

Using the same semiconductor laser technology as its entertainment cousin, one CD-ROM disk can easily contain 600 megabytes of data on one side. To put that number in perspective, if you were to download all the information on one full CD-ROM at 300 baud for 24 hours a day, it would take 184 days!

When the computer industry observed the way in which Compact Discs took off so suddenly — and, in turn, brought the cost of laser optical technology to incredibly low levels; $150 currently for a inexpensive CD player — they marveled at the possibilities for taking advantage of the new technique of storing digital data. The typical cost/performance ratio of optical versus magnetic media, even at this early stage of the game, is causing rapid changes in nearly every phase of electronics. If you are a part of the new digital audio scene on the professional level, or as a consumer, give yourself a pat on the back for helping bring this technology into everyone's hands.

The fine line between CD and CD-ROM is getting finer. Some Japanese companies have already announced combination CD-ROM and CD audio decks, to allow Compact Discs to be played on the same player that is connected to a computer for CD-ROM applications. In addition, Pioneer has announced a combination Laser Video/Compact Disc player, the Model CLD-900, which is capable of playing Laserdiscs digital audio, as well as Compact Discs.

How will this new technology affect the average recording engineer/producer, you might ask? Are we going to have to learn "all there is to know about computers" to take advantage of the optical revolution?

The answer is: Yes and No. It sure won't hurt to know your way around operating systems and computer languages, and to succumb to the fact that only software-based systems are capable of keeping up with rapid changes and upgrades. The software required to operate a virtual recording console, for example, would likely be quite complicated, involving large-scale simultaneous co-processing. However, indications are that large-scale storage media will bring to light greatly increased use of AI (Artificial Intelligence) and expert systems to

CD-ROM IN THE STUDIO

Compact Disc Read-Only Memory Applications for Data and Software Storage

by Bob Burr

make very complicated computer chores easily manageable by even the most illiterate go-fer on your staff.

Who's on First?

Several forward thinking companies in the professional audio arena are already embracing CD-ROM technology. Optical Media Services of Aptos, CA, has developed an extremely effective application of CD-ROM for the E-mu Systems Emulator II digital sampling keyboard, as described in an accompanying sidebar to this article. One Compact Disc can store as much data as 1,000 of the Emulator's 5¼-inch floppy disks. The OMS *Universe of Sound* CD-ROM disc, player and interface marks the first such product to be made available to the professional recording industry. It is only a matter of time before more digital recording applications of optical storage media emerge.

Another firm involved in early CD-ROM technology is Earth View, Inc., of Ashford, WA, which specializes in interactive applications for digital

— The Author —
Rob Burr is president of QL Digital Records QL Mobile Recording. A recording engineer/producer specializing in digital recording, and the manufacturing and marketing of Compact Discs, Burr is the system operator and a trustee of the Gold Coast Osborne Group, a computer club in the Miami, FL area. He has extensive knowledge of magnetic and optical media, and artificial intelligence, and frequently consults in advanced computer systems design.

audio recorded on CD-ROMs. Since CD-ROM can hold digital audio, digital data, and compressed, bit-mapped video graphics in any combination, the controlling computer can "talk"to you, show you pictures, maps or animation, run a program or access any data on the disk. A current example of the use of this multimedia concept is the newest generation of arcade computer games, such as *Dungeons and Dragons*.

Earth View's president, Brian Brewer, has provided the first demonstration of how all of these elements might be combined on CD-ROM, including holographic sound techniques, in a presentation that explains Compact Disc technology, titled *Interactive AudioMation*. Brewer uses IBM PC with a standard color graphics board interfaced to a Sony CDP-610 CD player.

Earth View is working in the area that promises to be the most effectively improved by CD-ROM technology: that of education. Interactive CD-ROMs will revolutionize the education of highly skilled, as well as moderately skilled personnel, by allowing each user to access knowledge to his or her own level of competence, and at a comfortable personal pace. Want to learn more about advanced electrical engineering? (Neither do I.) How about digital fiber-optic transmission systems? (Now you're talking.) Sit down and become proficient, at your own speed, using an interac-

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NEW TECHNOLOGY: CD-ROM in the Studio

situation is changing.

CD-ROMs will find their way into many of the current computer-assisted products found in our studios. One CD-ROM might contain everything from software that records changes in mix levels and equalizations, to those that help us keep track of clients, billing and studio time, and more.

We are seeing the embryonic stages of a new era of cost-effective data storage and retrieval. Compact Discs containing a combination of digital audio, graphics and data are beginning to emerge.

Firesign Theatre has recently released the first Compact Disc containing subcode graphics. The eight subcodes, labeled P through W in the signal, can carry additional information. The P subcode, for example, tells the CD player to mute the outputs when music is not present, and can be read by all CD players. Other subcodes are useful for indexing musical passages, or allowing instant access to a particular section of data. More often, manufacturers are including subcode outputs in late model CD players. In the case of the new Firesign disk, compressed graphics have been encoded into the Q subcode. So, if you have a player with a subcode output, and a subcode graphics demodulator, you can watch the pictures on your TV or monitor while you listen to the Compact Disc.

Where It's Going

Because of the ease of storing very large programs with "all the bells and

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AN OVERVIEW OF VARIOUS OPTICAL AND MAGNETO-OPTICAL STORAGE TECHNOLOGIES

Arthur C. Clarke was once quoted as saying that "Any sufficiently advanced technology is indistiguishable from magic." Since we live in the future now, what innovations in optical storage media and related fields lie ahead of us? Here are some acronyms to chew on:

Optical Media

• **CD-ROM: Compact Disc Read-Only Memory** is the heart of a new computer optical data storage system (Figure 1). CD-ROM is a single-sided, 120mm (4¾-inch) polycarbonate disk that utilizes a CLV (Constant Linear Velocity) data-storage method in one continuous spiral track. The data-stream rate remains constant, while the disc changes speed between 530 rpm near the center, to 200 rpm at the edge track. Throughout, averages of 150 kilobytes per second and seek times ranging from 0.25 to 2 seconds are common.

Each of the more than 270,000 data blocks of 2,352 bytes contains the following components: 12 bytes of sync; 3 bytes of address; 1 mode byte (which distinguishes the bit stream between data, sound, and graphics blocks, etc); 2,048 data bytes, and 288 bytes for error detection and correction (Figure 2). Track pitch is 1.6 microns with a 66mm footprint. Density equals 6×10^8 per square inch. The wavelength of the laser is 780 nanometers.

A binary "1" is represented by a land/pit or pit/land transition. The number of binary "zeros" is determined by the path length between these transitions (Figure 3). Nearly 100% of laser light is reflected from the land area, while the pit diffracts most of the light, reflecting approximately 30% of the light back to the sensor (Figure 4).

• **OROM: Optical Read-Only Memory** (or DataROM) is another standard that will find its way into industrial applications calling for data throughput speeds of greater than 1.5 megabytes per second. OROM uses a CAV (Constant Angular Velocity) storage scheme formatted into concentric circles, similar to standard floppy disks, allowing seek times of less than 50 milliseconds. Only one manufacturer is currently producing a player in this configuration.

• **LV-ROM: Laser Video Read-Only Memory** utilizes the same popular 12-inch, two-sided LaserVideo disk system available to consumers. Applications include personal interactive learning programs, such as flight training; legal and medical expert systems; and surrogate location scouting. Once again, it is the early success in the consumer market that makes this technology afforable to the computer industry. Many LV-ROM interactive systems are now integrating CD-ROMs to provide up to 31 hours of low-fidelity, compressed audio to their instructional system.

• **CV-ROM:** Yet another optical memory configuration, **Compact Video Read-Only Memory**, is basically a smaller version of LV-ROM that competes with the CD-ROM. Its major differences are CAV format using concentric circular tracks.

• **WORM: Write Once Read Mostly** is a disk that allows one write, after which it is not erasable — a perfect format for making small numbers of copies of CD-ROMs, without going through the entire injection molding process.

• **DRAW: Direct Read After Write** media allows us to record on optical disk (or card), and immediatly read the data. It is similar in many respects to WORM, but with different systems of formatting data. Some logistical problems occur when errors are introduced in the copy process.

• **EMOD: Erasable Magneto-Optical Disk** — sometimes referred to as Thermomagnetic or Optically-Assisted Magnetic —

Figure 1: Main Elements of a CD-ROM System.
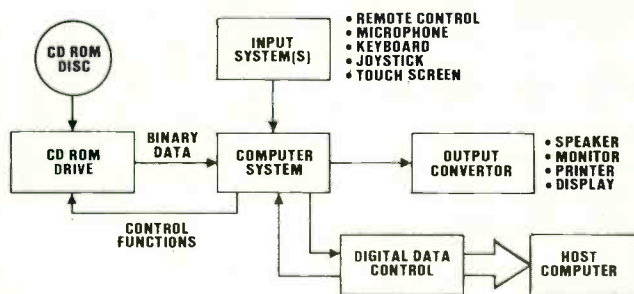
Figure 2: CD-ROM Data Block Format.

00	FF x 10	00	MINUTES	SECONDS	SECTOR	MODE	BINARY DATA	LAYERED ECC
SYNC (12 Bytes)			ID (4 Bytes)				DATA (2048 Bytes)	L. ECC (288 Bytes)

TOTAL: 2352 Bytes
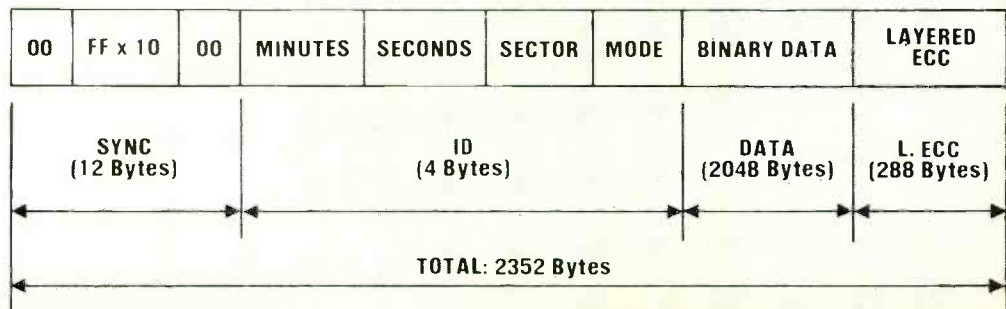
NEW TECHNOLOGY: CD-ROM in the Studio

whistles," CD-ROMs will allow the incorporation of artificial intelligence and inference engine systems to truly allow computers to be interactive with the user. In the near future, for example, a "smart" synthesizer could teach you music theory, test your ability to play various musical exercises, and suggest how long you need to practive. (As long as it doesn't tell me "door ajar.") In conjunction with sophisticated sound shaping, looping and sequencing software, sound samples can be infinitely processed, reprocessed, and stored on hard or floppy disk. Real-time, resolution graphics of music samples can help the brain correlate aural and visual components.

A fully automated virtual recording console could display any aspect of a signal waveform, spectrum analysis,

harmonic components, equalization and more. The issuing of regularly updated CD-ROMs containing massive programs to operate their consoles, will allow manufacturers to constantly offer more and more features to customers via an inexpensive medium.

A future multimedia multitrack machine may employ one EMOD (Erasable Magneto-Optical Disc) drive per audio or video track to give truly random access digital recording and playback of audio and video. Add a couple dozen digital microphones and a full blown digital virtual console and watch out! (First studio to get one, wins.)

These are merely a few obvious examples of how optical technology, and CD-ROMs in particular, will revolutionize not only the recording industry, but many facets of our lives. I am interested in your feedback concerning these issues, and how you feel this

might effect your aspect of the business. You can contact me through this magazine. ∎∎∎

AN OVERVIEW OF STORAGE TECHNOLOGIES — continued . . .

is a 130mm (5¼-inch) disk that has unlimited read, write and rewrite abilities. This system uses both a laser device and a magnetic read/write head. The sensitive layer of the disk is composed of an alloy film of rare-earth and transition metals, such as TbFeCo (terbium, iron, cobalt) or GdTbFe (gadolinium, terbium, iron). In its dormant state, the disk is unaffectd by the magnetic field or the read/write head. When the laser is focused on a point within the head's field, however, the laser beam heats the alloy to a point beyond its Curie point, where it is easily magnetized by even a small field. In the playback or read mode, the angle of polarization is rotated slightly providing modulation. One EMOD disk currently can store 300 to 500 megabytes, which is what Dr. David Davies has been working to perfect over at 3M's Optical Recording Project.

Watch for this format to eventually grab a larger share of the market, once production ramp-ups have occured. (However, EMOD will probably not achieve the low-cost factor of CD-ROMs multimillion per month production capacity.)

Data Format Error Correction

• **EFM: Eight-to-Fourteen Modulation** is an encoding scheme that converts eight-bit data to 14-bit data so that no consecutive "ones" can occur next to each other. The scheme allows a high bit density to be achieved without problems of resolution, to make the data self-clocking, and to minimize error propagation.

• **RBER: Raw Bit Error Rate** is a measurement of the number of errors in the media, expressed in exponential terms. Typically,

---

◄ **Figure 3: Compact Disc Data Recording Technique.**

▼ **Figure 4: Data Retrieval from Surface of Compact Disc.**

1.6 µm SPIRAL SPACING

0.6 µm SPIRAL WIDTH

PROTECTIVE LAYER

PIT
LANO SUBSTRATE
REFLECTIVE LAYER

DATA IMPRESSED INTO THE SUBSTRATE AS A SERIES OF PITS OF VARIABLE LENGTH ARE READ BY THE TRACKING LASER THROUGH THE TRANSPARENT SUBSTRATE LAYER.

PROTECTIVE LAYER
TRANSITION PATH LENGTH

00 1001 000001 0000 100000 10001000

A BINARY "ONE" IS REPRESENTED BY A LANO/PIT TRANSITION, WHILE THE NUMBER OF "ZEROS" IS DEFINED BY THE LENGTH BETWEEN TRANSITIONS.

RELFECTIVE ALUMINUM COATING
1.0 µm
0.11 µm
FLAT SECTION
PIT
PIT HEIGHT
CLEAR PLASTIC HAVING REFRACTIVE INDEX OF 1.5
LASER BEAM
1.2mm
ABOUT 1mm DIAMETER
BACK SIDE OF OISC

FLAT AREA
NEARLY 100% OF THE LASER LIGHT IS REFLECTED BACK TO PICKUP
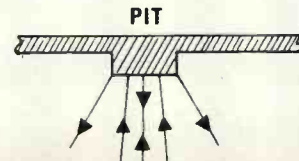
PIT
LIGHT IS DIFFRACTED SO THAT ONLY APPROXIMATELY 30% RETURNS TO PICKUP

"I can't imagine ever recording without BBE™ again."

Steve Levine, Producer of Culture Club, The Beach Boys.

Steve Levine's got an ear for hits. He can pick a winning song out of a hundred rough demos, take it into the studio and polish it till it's Triple Platinum. That's why he's topped the charts time after time with trendsetters like the Culture Club. That's why he was named 1984 British Phonographic Institute Producer of the Year. And

TYPICAL HARMONIC STRUCTURE STARTING TRANSIENT (UNAMPLIFIED)

TYPICAL HARMONIC STRUCTURE STARTING TRANSIENT (AMPLIFIED)

that's why Steve Levine records with the BBE 202R. When Steve produced The Beach Boys' new all-digital album, the 202R was on the team, too.

"BBE is to digital what equalizers were to analog. I'm particularly impressed with BBE's effect on synthesizers."

BBE has its own sharp ear for sound. It senses and instantly corrects problems in that all-important interface between amplifier and speaker. That's where phase and "overhang" distortion

develop, due to voice coil characteristics, reflected impedance from the environment, crossover impedance anomalies and the mechanical properties of dynamic speakers. The relationships among the fundamental frequencies, their leading harmonics and between the leading harmonics themselves become distorted in both amplitude and time. The result? Muddiness, poor imaging and pinched, colorized sound that lacks the presence and punch of the real thing.

The BBE 202R puts the clarity and sparkle back into amplified and recorded sound. We like to think of it as the "unprocessor." Rather than artifically altering the original source, BBE restores the natural harmonic balances that were present in the live performance. How? First it divides the audio spectrum into three bandwidths. Then it applies phase correction across the full spectrum and dynamic high frequency amplitude compensation as required. BBE's continual sampling of the mid/high frequency relationship allows this

correction to take place automatically. Convenient front-panel controls let you boost low frequencies and regulate the

SYSTEM PROCESS BLOCK DIAGRAM

amount of high-frequency amplitude correction to suit your needs. There's no encoding or decoding involved, so BBE can be used anywhere in the recording chain—from individual tracks on a multitrack tape to a mastering lab monitoring system.

Successful producers like Steve Levine count on BBE to bring that hit potential into focus. Why not discover the hidden potential in your own recordings?

To find out what the BBE 202R can do for your sound, contact your professional sound dealer.

Or write to us at Barcus-Berry Electronics, 5500 Bolsa Avenue, Huntington Beach, CA 92649, or call 1-714-897-6766.

BBE™ All the sound you've never heard.™ Barcus-Berry Electronics, Inc.

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AN OVERVIEW OF STORAGE TECHNOLOGIES — continued...

CD-ROMs have RBERs of 10^{-8} without error correction.
• **EDC and ECC: Error Detection** and **Error Correction Codes** facilitate compensation for minute imperfections in the production materials, those introduced in the production process, and minor scratches, fingerprints and dust on the disk surface, using CIRC (Cross Interleaved Reed-Soloman Code) to achieve its RBER of as little as 10^{-15}.

Small Computer System Interface

SCSI (pronounced "scuzzy") or Small Computer System Interface is a new intelligent interface designed for local-area computer networking of up to eight different peripheral devices (normally configured as one "host" connected to seven "slaves"), including the ability to interconnect more than one host for a type of "daisy-chaining" effect with peripherals such as a CD-ROM player, a MIDI-equipped digital keyboard, a laser or LCS printer, hard disk(s), a high-resolution video monitor, a voice box or D/A audio synthesizer, video digitizers and scanners, optical character readers, extra RAM (random-access memory), and so on.

Defined by some computer users as a "multi-ported bus," SCSI consists of a 50-pin port, of which normally 39 pins are used for data transfer and system handshake protocols. The main benefits of SCSI include the ability to transfer data in either a serial-to-serial or serial-to-parallel format. In addition, the speed of transmitting data, sent block by block in a synchronous mode (i.e., devices having constant timing intervals between successive bits, characters or events) to any device offering a SCSI port, can run up to 3 Mbytes per second. However, typical applications require an asynchronous transmission mode (for devices that possess different internal clock rates, for example), for transfer speeds which peak at 1.5 Mbytes per second. (Such rapid transfer rates via SCSI are in contrast to CD-ROM players, which currently are capable of downloading data at speeds between 150 and 175 Kbytes per second.

One possible solution might involve the provision of a RAM buffer capability within the CD-ROM player, into which is stored the required data. The host computer or system can then access the data at maximum SCSI transfer rates as it scans and interrogates in a cyclic manner, the peripherals attached to it, and so prevent data-transfer bottlenecks on the bus.)

Several CD-ROM drive manufactures currently are offering units fitted with a SCSI port. With intelligent software to drive the interface, installation of a SCSI controller board in the host computer will allow interfacing with any peripheral offering SCSI.

Interfaces and Peripherals

• **Laser Printers** use a rotating polygon mirror in addition to the drum and paper handling mechanisms required by all non-impact copier/printers. Because of the very clever adaptation of the Canon LPB-CX, plain-paper copier mechanism, or "engine," laser printers have taken an early lead in the page-printer race. Resolution is typically 300 by 300 dots per inch, and such devices are very instantly printing musical charts, lyrics, etc.
• **LCS: Liquid Crystal Shutter** printers are a new generation of non-impact printers utilizing a liquid-crystal shutter with 240 pixels per inch, similar to, but far less aliasing than, your wrist watch. Printing is accomplished using fiber optics and a florescent lamp to rapidly expose the print drum to changing characters on the liquid-crystal shutter. These are very quiet, fast (four pages per minute), compact, high-contrast printers suitable for printing to regular paper from optical media.
• **LED:** Similiar to LCS unit, **Light Emitting Diodes** and fiber optics are used to expose a photo-sensitive drum rolling across plain paper. The LED is much more complex than LCS, in that each pixel or point must be generated by a separate millimeter-sized light.

□□□

LAKE EXPANDS CAPABILITIES

What's new at LAKE? Besides the influx of new people . . . A host of new computer systems. Computers that assist in the design, engineering, drafting. and service of audio/video systems. One of the most exciting new computer systems is the audio departments Tecron TEF System 10. A portable audio spectrum analyzer that can be used in the field and the data brought back to the office for further analysis.

LAKE is involved in the design and building of television stations, recording studios, post production editing systems, and sound reinforcement systems worldwide. A computer system that could quickly analyze the acoustic parameters of any space was very important to the engineering department. They are currently using the TEF 10 to help expedite the engineering requirements of an expanding customer base.

An example of its value was recently discussed at a meeting I attended. It seems that microphones placed at a specific area on stage were experiencing excessive feedback. The client had tried a number of corrective measures to no avail. LAKE's engineers, using the TEF 10 were able to pinpoint the problem, something that at first

LAKE'S audio systems engineers Dennis Smyers (foreground) and Steve Blake analyze data on the TEF System 10

glance seemed insignificant, a steam pipe located near the speaker cluster was causing a strong reflection into the problem area. Covering the pipe with absorbent material, eliminated the problem.

Without a doubt, this type of commitment on the part of LAKE in R & D, positions them as the systems company of choice in the audio field. Contact them at (617) 244-6881.

DIGITAL SOUND
FOR MOTION PICTURES

PART TWO: POST PRODUCTION AND THEATRICAL PLAYBACK

Part One of this article, published in the October 1985 issue, covered the areas of production recording and random-access editing/mixing. This final part will discuss the use of digital sound recording in post-production and for theatrical exhibition.

by Larry Blake

It appears that "hybrid" will continue for many years as the operative word describing the use of digital sound recording technology in motion pictures. The prevalent attitude seems to be that movies sound pretty good with today's technology, so why change? Hollywood, the Land of Stationary Inertia, has many working habits and even more money invested in post-production techniques based on 35mm analog mag. Also, the multitude of options presented by the new world of digital sound recording — digital multitrack or mag film; optical disks or hard-disk storage, etc. — will further delay the time when digital motion picture sound can be dealt with as easily as analog mag is today. No studio wants to convert its plant to the Acme XYZ digital format, only to find itself with obsolete equipment on the "Digital Road Not Taken."

This situation is in contrast to the music-recording world, where all-dig-

ital albums are commonplace, and the Compact Disc format completes the final link to the consumer. The path to the moviegoer, however, is a much more rocky one.

The first tough decision for intrepid filmmakers trying to create an "all-digital" soundtrack is how to get edited sound elements to the dubbing stage. The most obvious solution is to lay digital two-track sound effects in sync onto a multitrack digital recorder, with the multitrack tape going to the re-recording stage. The draw-

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MASTER RECORDER A80VU MKIV
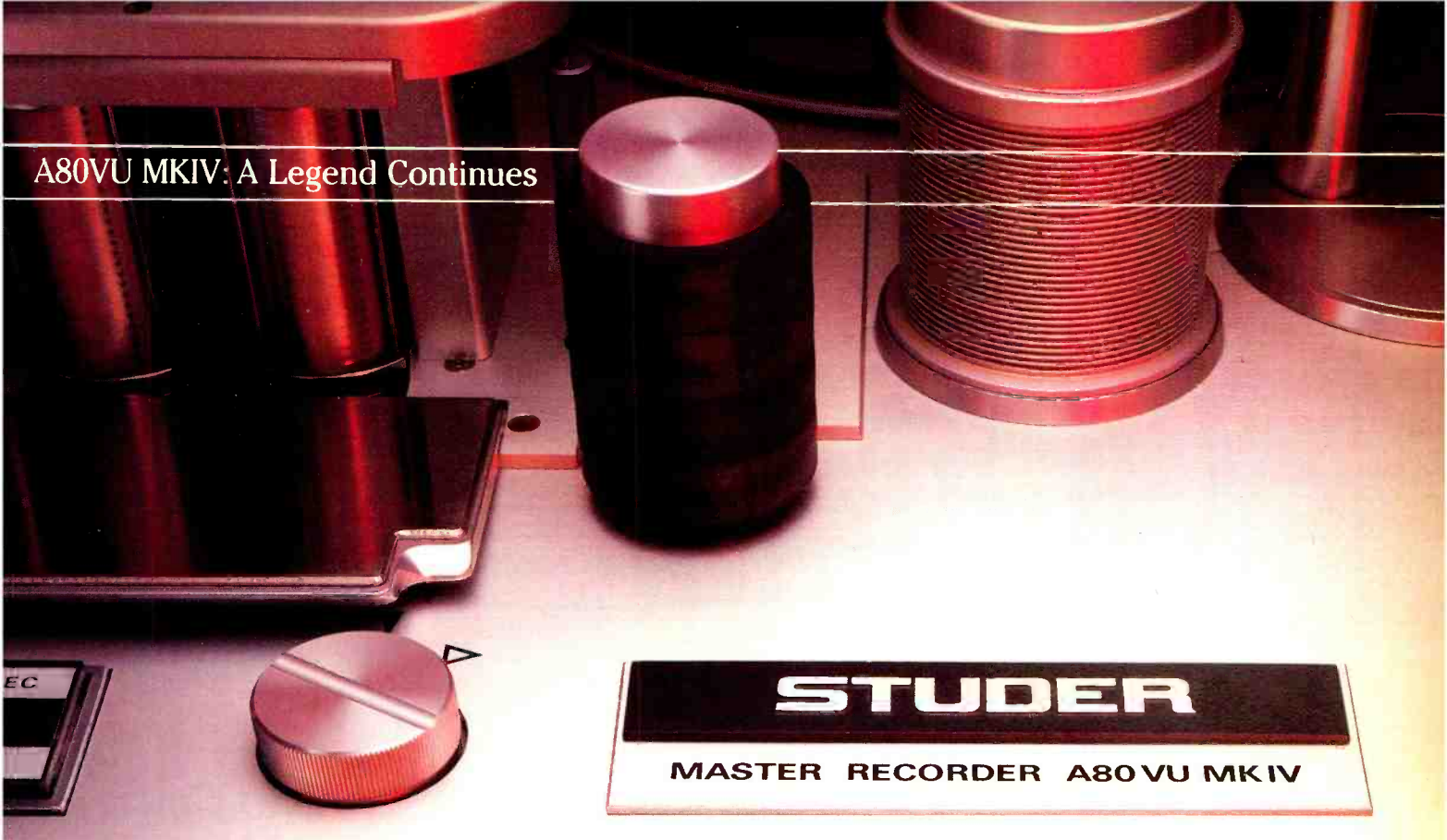
This is our newest multitrack. It is also the most affordable multitrack in Studer history.

For the fourth time since its inception, we've changed the A80VU. We've improved the sonic performance, tape handling, and durability. And we've substantially lowered the price.

Same outside, changes inside. In keeping with the Studer tradition, we made no superfluous cosmetic changes. We're not going to tell you this is an "all new" recorder. It isn't. It is a proven, legendary recorder incorporating several significant improvements.

Uh-oh, Something Is Missing. Yes. The transformers are gone. They've been replaced in the input and output stages with new high performance active balancing circuitry. Other MKIV improvements include a new master bias oscillator, extended record headroom, and a new record and bias driver compatible with all present and future high-bias requirements. Record electronics are now fully compatible with Dolby HX Pro* requirements.

Smoother Shuttling, Hardier Heads. The MKIV's new tape tension control system provides smoother tape handling, while a new extended wear alloy for record and play heads greatly increases head life.

Never Lower. The list price of the A80VU MKIV 24-track is lower than any of its predecessors. And that's in straight dollar figures, without adjusting for inflation. What's more, the A80VU MKIV now has a list price lower than most of its competition.

No Hocus-Pocus. How could we make the A80VU MKIV better and lower the price at the same time? Simple. We make it in Switzerland, and you pay for it in dollars. The favorable exchange rate does the trick. That means you get advanced electronics, Swiss precision, and low price. If you act now. This can't go on forever.

Your Time Has Come. If you've always wanted a new Studer multitrack but thought you couldn't afford one, your time has finally come. Call today and find out why the A80VU MKIV is one of the most advanced recorders available at any price. And then ask about our new lower prices. Be prepared for a pleasant surprise.

For more information, call or write: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

*Dolby HX Pro is a trademark of Dolby Laboratories.

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www.americanradiohistory.com

Update on Motion Picture Sound Systems: The Battle of the Acronyms

Since its first sales in 1982, the JBL Model 4675A direct-radiator theater loudspeaker has virtually eclipsed the horn-loaded Altec Lansing A-4 Voice of the Theatre as the *de facto* industry standard loudspeaker for motion picture reproduction by virtue of the imprimatur of its ubiquitous presence on re-recording stages and screening rooms, not to mention its use in over 1,500 commercial theaters in the U.S.

During the past three years, the Model 4675A has been installed in over 12 re-recording stages, including Sprocket Systems (Lucasfilm), Warner Hollywood (Stages A and D), The Burbank Studios (Dubbing 1-5), Glen Glenn (Stage S and Dubbing 1, with three more to follow), Todd-AO Stage B and MGM Cary Grant Theater (Main Theater). In April 1984, the Academy of Motion Picture Arts and Sciences installed a bi-amped Model 4675A system in its Samuel Goldwyn Theater, the first time the Academy has not used the Voice of the Theatre since the latter's introduction almost 40 years ago.

Ken Lopez, VP of sales for JBL/UREI theater products, says that the primary benefit of the Model 4675A, compared to standard theater speakers, is "the extremely smooth power response over the frequency range from about 100 Hz to 8 kHz. This means that minimal equalization is required to meet the desired wide-range [X] curve used in theaters today."

Half of the theaters named above employ active crossovers and bi-amping, using either the Lucasfilm THX or JBL Model 5234A network.

The THX crossover forms the heart of the sound system that Lucasfilm has been licensing to theaters since the May 1983 release of *Return of the Jedi*; as of January 1986, the system has been installed in almost 60 theaters in the U.S. The THX system not only includes installation of the sound system, but also a comprehensive program of acoustics and projection standards which is inspected by the Lucasfilm staff every six months. [See the December 1983 issue of *R-e/p* for a lengthy look at the design of the THX System —Editor.] Contrary to a widely held misconception, the THX Sound System has nothing to do with the recording of a film soundtrack, nor does it in any way replace Dolby Stereo.

The JBL Model 4675A, which is currently the only theater system approved for use in THX systems, was designed in 1981 by Mark Engebretson and John Eargle. It is comprised of two 2225J 15-inch woofers mounted in a 4508 vented box enclosure, with a 2445J compression driver in a 2360 Bi-Radial™ horn. Smaller theaters with little space behind the screen will use the Model 4670, which is identical to the Model 4675A except for the smaller 2380 horn on a 2445J HF driver. Mounting the behind-the-screen speakers flush in a wall —as can be seen from the accompanying photograph of the Academy Theater — has become standard practice, and serves to smooth out the system's low-frequency response by providing boundary reinforcement. (It should be noted that this "two-pi" mounting is also very effective with standard Altec and other horn-loaded loudspeakers.)

Theater speakers made by two other major manufacturers are currently under consideration for use as part of the THX System. The only approved surround speaker is the Boston Acoustics A70T, although units from Electro-Voice, Altec and JBL are presently being tested.

Clyde McKinney, director of the THX program at Lucasfilm, is pleased with the latest generation of 250-watts-per-channel amplifiers, and counts the Ashly FET-500, BGW 8000, Hafler DH-500, UREI 6300 and JBL/UREI 6260 among the approved devices. He says that the requirements of the THX System "are not so stringent, but we are concerned about not only the power and frequency response, but also how they clip."

A stereo amp is used for each behind-the-screen speaker, with equal power to the woofer

The Samuel Goldwyn Theater sound system at the Academy of Motion Pictures Arts & Sciences untilizes five, two-way JBL Model 4675 A-2 cabinets behind the screen, and eight Model 4446 subwoofers. The units are biamped with BGW 250 and 700s.
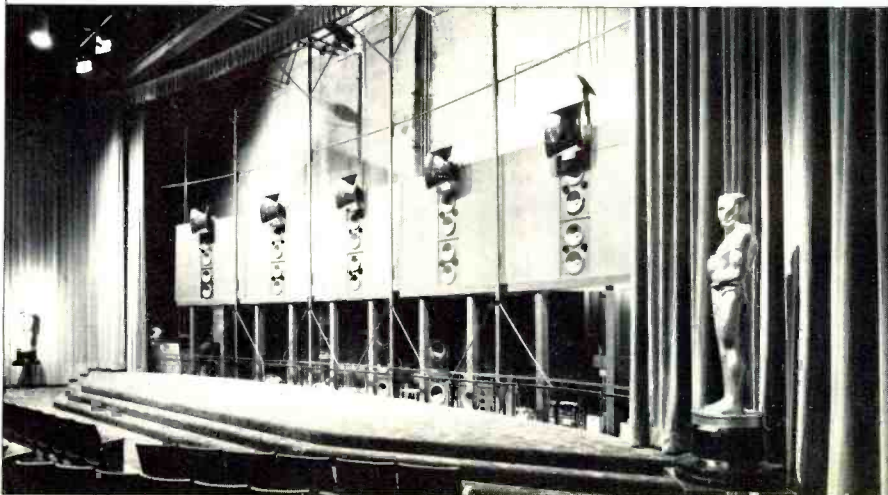
DIGITAL FILM SOUND

back with such a system is simply one of cost, since anywhere from $125,000 to $200,000 in digital equipment (depending upon system design) is tied up on each editing station.

Also, the multitrack medium requires time-consuming offsets to slip a track while dubbing or conforming to picture changes; with standard 35mm mag, an individual playback dubber is quickly advanced or retarded to the required number of frames or sprocket holes. Picture changes in the latter stages of post-production are accommodated by conforming, on an editing bench, the sound elements to match the latest version. Effects recorded on multitrack tape would need to be transferred machine-to-machine: a very expensive proposition. (Of course, digital mixing from random-access hard disks solves the offset and picture change problems, among many others. The benefits of random-access editing and mixing were discussed in Part One of this article, published in the October 1985 issue of *R-e/p*.)

Where music recording requires the use of just one or, at the most, three multitrack recorders, subsequently mixed to a two-track stereo master, digital stereo motion picture re-recording would require at least two, and possibly as many as five, digital multitracks during the final stereo mix, where separate four- or six-track dialog, music and sound effects "stems" are recorded. With digital multitracks costing from $113,000 (Sony PCM-3324 24-track) to $154,000 (Mitsubishi X-850 32-track), almost a half million dollars is needed to equip such a dubbing stage.

Furthermore, there is currently no field-tested digital theater reproduction system. The final stumbling block here, as has been the case with every improvement in film sound, is that the sound should be on the same piece of film as the image. Projecting separate picture and track — "double system" in film jargon — is a chancy proposition no matter how "bulletproof" the system.

The following is a look at the current attempts to overcome the obstacles, technological and otherwise, that stand in the way of all-digital soundtracks becoming an everyday reality.

Digital Magnetic Film

The last big change in film sound occurred 35 years ago with the advent of professional magnetic recording. The transition was relatively painless and inexpensive for film studios, since the same transports that had

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been used for 20 years to mix optical soundtracks remained in use with just the addition of new electronics and magnetic heads. Likewise, editing equipment and mixing consoles required little or no modification.

The retrofitting idea is being carried over with the next leap in motion picture sound, to digital recording, with attempts to record digital sound on sprocketed 35mm mag. It would be something of a testament to the inertia of standardization in the motion picture industry if an ancient dubber that originally played optical tracks in the early-Fifties was converted to play digital mag in the mid-Eighties!

Almost all of the problems noted at the beginning of this article — cutting digital tracks without spending a fortune, getting them to the dubbing stage, and slipping sync when there —can be solved by using 35mm *digital* mag for editing.

Magna-Tech Electronics, Inc. of New York, the largest manufacturer of magnetic film recorders, is currently involved in research on 35mm digital mag. Bob Ebernz, Magna-Tech vice president, says that they are in the early stages of development, with no delivery date set. He also said that they will adhere to the DASH recording standard in their system.

In March 1985, Digital Entertainment Corporation, a wholly owned subsidiary of Mitsubishi Electric of Japan, purchased the assets of Quad Eight/Westrex, thus bringing together Quad Eight, one of two major film dubbing console manufacturers, and Westrex, which has manufactured film recorders for over 55 years, with the experience that Mitsubishi Electric has with digital recording.

Cary Fischer, director of marketing at the Mitsubishi Pro Audio Group, says that the company is currently investigating the use of the Prodigital (PD) format on the sprocketed 35mm medium. (PD is a new stationary-head digital tape standard agreed upon by Mitsubishi, Otari and AEG Aktiengesellschaft — formerly AEG Telefunken — for two- and 32-track recordings. Other PD tape formats, such as 16 tracks on ½-inch tape and eight tracks on ¼-inch, are expected to follow.)

Considered below are some of the issues regarding the use of digital 35mm mag:

• There is no disagreement that whatever form digital 35mm takes — DASH, PD, or ? — the mag has to be read by standard analog Moviolas, sync blocks and dubbers. This means that an analog guide track must be recorded in the same place as the 200-mil track on 35mm stripe, the same position as track #1 on 35mm three-track.

With 35mm digital mag, the bottleneck between digital sound effects-production tracks and the film dubbing stage would be opened with minimal cost to the producer. Only the digital 35mm mag channels that transfer sound effects, production track reprints, Foley and ADR would have to know that digital sound is being used in the film; the editors would not be aware of it *per se*, because they would cut to the analog guide track using standard equipment. The re-recording stage would, of course, have to retrofit the dubbers with new digital replay electronics and head stacks.

It has almost been assumed by

MOTION PICTURE SOUND — continued ...

with equal power to the woofer and the compression driver because, as THX designer Tom Holman states in the system's instruction manual, "amplifier clipping cannot be considered on an average basis, but on a peak basis. There should be enough power available even during a cymbal crash to play undistortedly."

Standard equipment in THX and other top 70mm installations are Teccon Sendust "hybrid" magnetic heads. Since first making heads for 35mm mag recorders, Teccon heads have become standard equipment on 35mm mag recorders and playback dubbers, with many studios claiming that a headstack can last over 10 times that of standard heads. John Bonner, chief engineer at Warner Hollywood Studios, says that when he first installed a

Klipsch TMCM four-way cabinet as used in the HPS-4000XL systems.

70mm printing channel, he went through one 70mm erase/record/playback headstack in a few months; the Teccon replacement lasted more than five times longer. In addition, he considers that the "contour effect with head bumps is minimized and the gap scatter is almost immeasurable. On other heads it is *very* measurable."

This reputation for quality has led to Teccon heads being used in commercial theaters and, in fact, are required for all 70mm THX theaters. Because of the tight head wrap of 70mm film through the "penthouse" containing the mag headstack, six weeks is all it takes, when running matinees, to wear out a standard laminated 70mm headstack.

HPS-4000XL Theater System

One of the opposing camps in the "Battle of the Motion Picture Theater Systems" is the HPS-4000 system, which was designed by John Allen of Boston, and is composed of Klipsch Theater Speakers. Klipsch and Associates of Hope, Arkansas is familiar to most *R-e/p* readers for its corner-placed Klipschorn home speakers. Allen's top-of-the-line HPS-4000XL four-way system, like all Klipsch theater speakers, is fully horn-loaded. He notes that with full horn loading "the back air chamber is sealed tight and the drivers cannot bottom [i.e., reach the end of their excursion]. The amount that the driver has to move back and forth is reduced by about 90%. With this you get a corresponding reduction in distortion: you increase your efficiency by two, and you cut the distortion roughly in half. The standard complaints about the Altec A-4 and other ported box speakers do not apply."

Allen recommends bi-amping (quad-amping?) HPS-4000 systems only in the very largest theaters because of the similarity of efficiency between woofer, mid-range and tweeters. As a result, one of his biggest selling points of Klipsch speakers is their efficiency (109 dB at one watt/one meter) and the use of one amp per speaker channel in lieu of bi-amping.

Allen has installed three-way HPS-400 systems in 120 theaters and, to date, the four-way HPS-4000 XL system has been installed in only two theaters, Plitt's Century Plaza II in Los Angeles and the Avalon in Washington D.C. The HPS-4000 system is not currently in use in any re-recording stages.

Electro-Voice has recently assembled systems for cinema applications that are similar to JBL Model 4675A. The TS9040D system has the TL606DW low-frequency unit with two 15-inch woofers; DH1 HF driver into HP9040 horn, and the XEQ504 passive crossover. The XEQ2 electronic crossover is available for use in bi-amped systems, and provides time delay for the LF output for phase coherence with the HF driver.

The EV TS940 systems utilize the smaller HP940 horn and XEQ804 passive crossovers.

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DIGITAL FILM SOUND

many that if digital 35mm mag does come, it will be retrofitted onto existing machines at minimal cost. Many close observers have questioned the ability of sprocketed transports to maintain the stability necessary for digital reproduction, and the use of a sprocketless dubber has been discussed. On this matter both Magna-Tech and Mitsubishi have "no comment."

While digital 35mm mag will allow the same equipment to be used during editing, sound editors will have to give up one of the big advantages when working with mag: the ability to scrape the oxide of tracks to remove noises and to create fade-ins and -outs. Digital razor-blade editing is one thing; razor-blade *de-essing* might be pretty tough.

• If one assumes that digital multitracks would be used to record the three- to six-track pre-mixes and final mixes during re-recording, digital 35mm mag can be seen simply as a way to get cut digital mag tracks to the dubbing stage, and only two or three tracks might be needed.

Mitsubishi's Cary Fischer says that the company's first prototype 35mm digital mag machines will be available in numerous track configurations, not unlike the current situation with analog mag.

• Regarding the oxide formulation to be used with the digital mag recorders, Fischer says that while they have had "encouraging results with existing [35mm mag] stocks, we are investigating a lot of different avenues." Magna-Tech's VP Bob Ebernz says that they have had successful results with videotape formulations, which of course, are used on multitrack digital recorders. The capability of videotape to withstand the rugged world of sprocketed film, however, remains to be seen.

Multitrack Digital Recorders

The small number of films that have used digital multitracks during the re-recording process is suprising for two reasons. Not only is there an increase in sound quality gained by avoiding three mag-film generations — pre-mixes, final mix, and printing masters — there is a practical reason for the industry to use such machines: All of the final sound elements from the pre-mix stage onward would be contained within a few reels, considering the 60-minute running time of the two major multitrack digital formats. Also, in a few hours, with tape costs of less than $600, the stems and printing masters could have a digital copy in either PD or DASH format, all
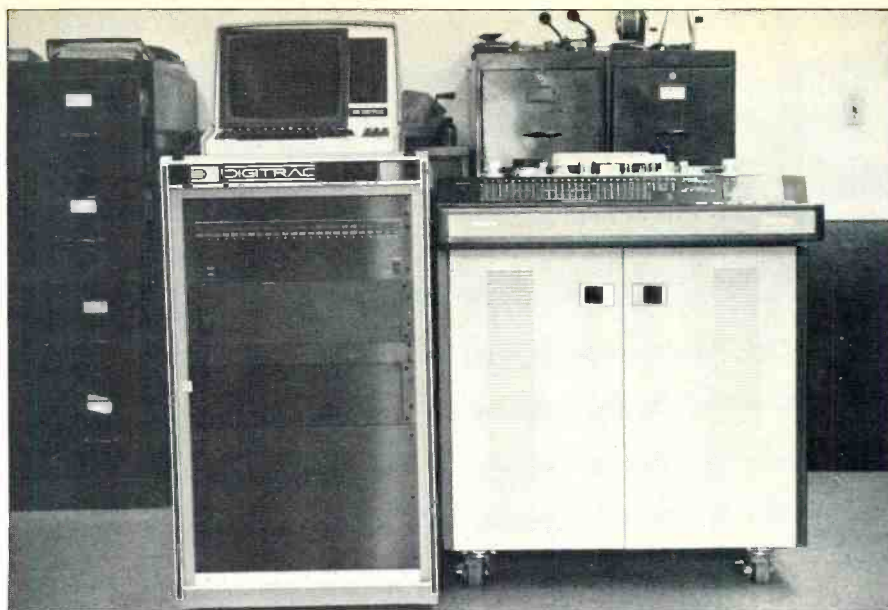
The Glen Glenn Digitrac Control Tower for controlling the synchronization of digital multitracks with a conventional film replay chain.

on a few inches of shelf space. For reasons of cost and trouble, usually only the printing masters are ever "copy protected."

In the final tally, a producer would save over $20,000 in tape costs by using digital multitracks during re-recording — not to mention that the digital masters would take up less than a foot of shelf space, compared to 25 feet with mag film.

• If digital multitracks are used at the dubbing stage strictly to record the pre-mixes, final stems and printing masters, then sound would be cut on either analog or digital 35mm mag. Thus the flexibility to shift individual tracks during the pre-mix would still be retained.

• Major re-recording stages have anywhere from 15 to over 40 35mm playback dubbers in their machine rooms, along with two to four 35mm recorders. During pre-mixes and final mixes of "busy" reels, it is not uncommon to have as many as 70 playback channels being mixed onto as few as three, or as many as 18 tracks, as is the case of a six-track final mix with separate dialog, music and effects stems.

Because of this variable demand for separate tracks, probably no less than three digital multitracks will be needed to playback the pre-mixes during final re-recording. A likely scenario: One would contain the music, another the dialog and some sound effects, and the third sound effects only. All three machines would have four to eight blank tracks left open for recording offset LCRS pre-mixes from other multitracks; a two-track DASH or PD recorder would not have enough tracks.

All of which brings up the concept

that a lower-cost, playback-only version of PD and DASH multitracks might make the transition to digital more affordable for dubbing stages. However, neither Sony nor Mitsubishi forsees manufacturing a playback-only multitrack unless the demand is present.

• As long as multiple sound elements are recorded on a serial medium like multitrack tape, there will remain the problems of re-cutting the sound to match picture changes. (Operationally, this applies equally to analog or digital, although with digital, of course, no generation loss occurs.)

• Having eliminated three or more analog generations, the quality of the final Beta and VHS Hi-Fi videocassettes and digital/FM Laser-Vision discs should be much improved. Interlock duplication from a digital master, either on PCM-1630-encoded one-inch videotape, or from half-inch VCR with dbx Model 700 or JVC VP-101 encoding, for examples, allows the quality to be preserved without having to use the sound on the one-inch analog videotape.

Glen Glenn's "Digitrac": An All-Digital Post-Production System

Glen Glenn Sound's involvement in digital recording for motion pictures began with its work on the short *Digital Dream* in late 1983. Since that time, research in the application of digital sound in all aspects of motion picture sound recording has been ongoing, with special attention being paid to use of the Sony PCM-3324 digital multitrack in post-production. The research team is headed by Glen Glenn chief engineer Dana Wood, with software design by Walter Baker

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and hardware modifications by Todd Boswell.

The company is now actively marketing, for use by its clients, what it considers a complete system for editing, mixing and exhibiting all-digital feature motion pictures. This Digitrac includes sound editing with Glen Glenn's PAP (Post Audio Processing) system which, in its standard form, allows for sound effects to be layed onto a 24-track analog recorder from broadcast carts or ¼-inch tape. The digital version will utilize either ¾-inch PCM-1630 or half-inch EIAJ-format tapes as source material to be recorded onto an interlocked -3324. Pre-mixing and final dubbing will employ from three to five -3324 multitracks, with one serving as the master recorder, and the others working as very expensive playback dubbers.

The "black box" (actually it's blue) that forms the heart of the Digitrac system during re-recording controls synchronization, track routing and communication from the re-recording console's motion control, record on/off and selsyn/direct switches, so that "the mixer thinks he has a Magna-Tech on his hands," says Glen Glenn's Dana Wood. "During the mix, as far as the client and mixers are concerned, it is a standard analog movie."

Lockup time between the digital multitracks and the film chain (in this case represented only by the high-speed projector; no 35mm element containing SMPTE timecode is required for film chain/SMPTE sync) is claimed to be less than two seconds.

The Digitrac "tower" will remain in the projection booth next to the digital multitracks, and in its current form will allow the Sony 3324 to be interfaced with the ADM dubbing consoles and Magna-Tech-equipped film chains at Glen Glenn. However, Wood emphasizes that the box can be easily adapted, by plugging in interface "personality" cards, to work with any multitrack digital or analog recorder and any film chain and dubbing console combination.

PAP editing rooms at the Glen Glenn facility in Hollywood can also be used in what the company refers to as the "configuration" process — i.e., conforming -3324 24-track cut effects elements, pre-mixes or final mixes to match picture changes. Executing the offsets "off-line" in a PAP room will cost the producer $150 an hour instead of over $700 "on-line" in a dubbing stage.

As envisioned by Glen Glenn, a producer wishing to have an all-digital movie would need three multitrack digital recorders through the post-production sound process prior to re-recording: two for PAP sound editing and one for ADR/Foley recording. Two of the machines could be used in a PAP room during off-hours to conform already-cut reels to picture changes by transferring, with offsets, in the digital domain between two -3324s.

During the final dub, Glen Glenn VP Rick Larson anticipates having two multitrack digital machines in a PAP room devoted to configuration. "It's nice to say that you could do the conforming during the graveyard shift, and use the same machines that are on the dubbing stage, but directors don't want to wait. They want things *now*."

The anticipated surcharge for utilizing the digital system during dubbing, which would require at least three, and possibly up to five -3324s, is expected to be in the range of double today's cost of a stereo dub. With four weeks on the dub stage, the all-digital status might cost the producer an extra $100,000, not taking into account the cost of the machines' use during sound editing. (There goes the $20,000 savings in mag stock!)

Larson notes that Glen Glenn is prepared to do their first all-digital feature at no extra charge to the client. The company hopes that such a film will show other producers that the Digitrac system is indeed a viable proposition.

Digital Music Recording

As might be expected, multitrack digital recorders have seen their most extensive use during film post-production in the recording of music scores. Since 1979, probably dozens of films have had their music recorded digitally, although only a few of them —the re-scored *Fantasia, Digital Dream, Metropolis*, and *Home of the Brave* [the latter being Laurie Anderson's concert film, which this author will discuss in a subsequent issue — *Editor*.] — have stayed digital to the final print master. Instead, digital film scores have utilized, for the most part, 35mm mag mixdowns for all editing and mixing. (Similar techniques with digital multitracks have been used in post-production of many Music Video specials, including *Stevie Wonder Comes Home* and *Frank Zappa In Concert*.)

Among the scoring stages that regularly use digital multitracks in film and TV scoring are: CTS in London, which combines a Sony PCM-3324 with the first Neve DSP digital console; and The Burbank Studios, CA, which has had a Mitsubishi X-800 32-track since Spring 1983, and was the

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scene of many early digital scores with the 3M DMS.

(So far, TBS has used the X-800 32-track only once during final dubbing, for *Body Double* in October 1984. [See the October 1985 issue of *R-e/p* for a description of this process — *Editor*.] Since that time, the Post-Production Sound engineering staff at TBS has built an interface box, which allows entire control of the X-800 from the facility's Quad-Eight re-recording consoles. Thus, the dialog, music and effects mixers will have individual control over their stereo stems. TBS will soon take delivery on two additional 32-track Mitsubishi machines.)

During most orchestral film-scoring sessions, where everything is recorded in one pass and no overdubbing is planned, the three-track monitor mix is recorded on 35mm mag, and is intended for use during the final mix. A backup multitrack — analog or digital — is required in case re-mixing of the score is necessary, or to mix the album master. Two Los Angeles-based scoring mixers, Bob Fernandez of The Burbank Studios, and independent mixer Bruce Botnick, both use the digital format to record the monitor mix.

Fernandez, who has tracked to dig-

Two interlocked Sony PCM-3324s at Samuel Goldwyn Theater for digital playback of Giorgio Moroder's re-issue of *Metropolis*.

ital the scores for *Pale Rider, Pee Wee's Big Adventure, Ghostbusters,* among others, records his three-track monitor mix on the X-800, leaving 29 tracks for the separation of individual instruments. The custom Quad-Eight console at TBS Scoring 1, like most purpose-built film-scoring boards, has a separate film monitoring/recording section. The tape returns from the Mitsubishi X-800 come up at this

point, allowing Fernandez to punch in on the three-track mix knowing that the levels will match. He says that this technique often saves money, because "instead of doing a separate take, the composer or conductor can often get the correct balance by changing the level of an instrument and punching in on one track of the monitor mix. Normally it would be back to the top, and do it all over

again." Fernandez is also able to save money by transferring only the "print" takes to 35mm mag for editing.

Fernandez finds the 32-track format an ideal storage medium: "Not only do you have your multitrack layout, plus your finished [LCR] film mix, but you can also dedicate two tracks for your stereo [album] mix. It makes everything nice and neat."

Jim Walker, scoring maintenance engineer at TBS, says that "the digital multitrack is one of the most reliable tape machines that we've ever used. Analog parameters such as print-through, biasing, and EQ and level calibrations are no longer something we have to deal with on a daily basis. In two years, there has been virtually no mechanical or electrical problems with the X-800."

Botnick has recorded digitally the scores for such films as *E.T. The Extra-Terrestrial*, *Poltergeist* and *The Color Purple*. Since 1980, he has recorded the monitor mix of his scores on a pair of synchronized Sony PCM-1610 digital processors and companion ¾-inch U-Matic VCRs. One -1610 handles left and right tracks, with center and surround being recorded on the second -1610.

When Botnick has access to a digital multitrack, he will record his four-track (LCRS) film mix on it, and will only use one 1610 to record the two-track mix for album release. He has recorded with both the Mitsubishi X-800 and Sony PCM-3324 transports, and owns one of the latter machines.

If two two-track -1610 tapes are later used for the two-track album master, access to center speaker information provides Botnick with the ability to do minor rebalancing without having to go back to the multitrack master. "Left and right are at 'zero,' and center is down 3 dB [on the two-track record mix]," he explains. "Any mix that you hear on a three-channel [LCR] format should conform exactly to a two-channel balance. I have found that 10 out of 10 times, it does."

Botnick is careful to have each monitor speaker set to the correct Dolby playback level and wide-range curve, in addition to listening through the Dolby 4-2-4 matrix, so that he is aware of the effect of the Dolby Stereo encoding process.
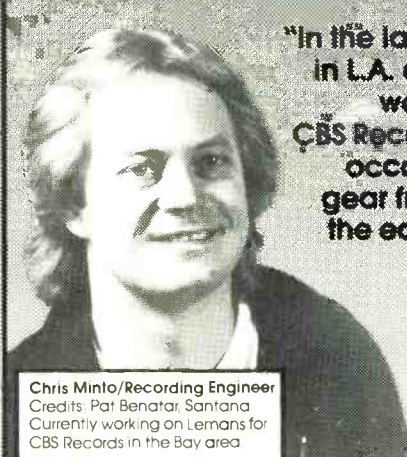
Botnick always simultaneously records the LCRS mix to 35mm four-track mag, and to date this film is what has been heard by the public — i.e., none of his films have utilized digital machines during re-recording. He emphasizes that 35mm mag is not too shabby of a recording medium itself: "I think it's the best analog medium I've ever heard. Because the tracks are so wide and the oxide is so thick [3

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or 5 mils], it's so quiet that it gives digital a strong run for its money."

Digital Film Re-Recording Consoles

One area of digital recording that, so far, has not attracted a great deal of interest is digital consoles. Part of this phenomenon is undoubtedly because they are so new — Neve DSP board at CTS in London went on line only early last year. Another reason is that many millions of dollars have been spent in the past few years installing film boards, with Quad-Eight and Harrison cornering the bulk of the U.S. market.

Given the more than "digital-ready" specifications of these boards, perhaps the change to digital consoles will have to wait until they form an integral part of a disk-based system. (While some might argue that it is undesirable to go through so many A-to-D/D-to-A conversions between digital recorders and analog consoles, it is probably safe to say that the reproduction systems in most theaters render such minor quality losses a non-issue.

Cary Fischer of the Mitsubishi Pro Audio Group says that "it is our goal to march forward into full digital consoles, in addition to digital control of analog consoles."

Digital Audio in Theaters

The perennial problem facing film-sound engineers since the beginning of time — (1926 that is, with the introduction of sound films) — is that, while the soundtrack preferably should be on the same piece of film as the picture, each new innovation in film sound technology seems to preclude such "single-system" projection, if only temporarily.

The Jazz Singer and many early sound films were released on inter-locked 33-1/3 rpm Vitaphone disks, since the art of electronic recording on disk — though only a few years old —was still superior to optical sound recording of the day. Very soon, however, editorial (how do you edit a disk?) and logistical problems of the Vitaphone system led the Hollywood sound community to adopt optical recording in both the studio and on composite release prints.

Before the coming of magnetic sound of Hollywood during the late Forties, double-system projection with optical soundtracks would be used rarely, most notably for the legendary original roadshow engagements of *Fantasia* in 1940. The four-track optical reproducer was interlocked with the projectors in only 14 theaters.

The next technical advance occurred over 10 years later, with the premiere of *This Is Cinerama* in the Fall of 1952. Double-system sound, using a seven-track mag film reproducer, was not a bothersome design problem with Cinerama process, since it already had to interlock three projectors.

Three-track 35mm mag interlock was soon adopted by Hollywood for stereo presentation of standard and 3-D films. This system was used for less than a year, however, when the CinemaScope format — with four magnetic stripes on a 35mm print — was introduced for *The Robe* in Sep-

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Who's On First?
The Misrepresentation of Digital Sound in Films
A personal commentary by Larry Blake

Over the past few years, with the gradual introduction of digital recording technology in motion pictures, there has been something of a modern "digital rush" taking place in California.

This is the basic scenario: Someone uses a digital recorder at some point during the post-production phase of a film, possibly resulting in a soundtrack that is somewhat quieter. So far, so good. Then, when the film is released, there is a flurry of publicity regarding the use of those digital machines. Adjectives like "first," "best" and "only" are bandied about freely.

Ultimately, these statements result in confusion on the part of the public, which does not know enough to separate truth from hype. Those in the industry react with a combination of derisive laughter and anger, the latter especially from those who think that *they* were the first to . . .

Possibly the only two things that *really* matter in regard to a film's acceptance are how much money it makes, and how good it is.

In the opinion of this writer, digital sound recording, and film technology in general, have no effect *whatsoever* on box office grosses. For example, it seems that every spring Hollywood trade papers run articles about the impact 70mm presentation has on the audience, and how the big "Seven-Zero" on a marquee lures the public out of their living rooms. These studies talk of how people prefer 70mm projection, which might be true. However, none of them ask the public if they would see a film primarily because it was exhibited in 70mm? The answer would have to be "no": they just have some vague idea that 70mm is "better." But I'd like to know how does that justify the expense of 70mm prints to the distributors?

Regardless of whether a film is a hit or a flop, or is good or bad according to critics, greatness cannot be quantified. Test equipment, Academy Awards, calendars ("the first . . ."), or good reviews, in and of themselves, don't make a film or its soundtrack worthy of attention. Either a film is good or it isn't, to the people who buy the tickets, and either it makes money or it doesn't. Digital sound recording, or any technological improvement for that matter, in this writer's opinion, has *nothing* to do with either.

There is a term in anthropology known as "parallel invention," meaning the simultaneous discovery by two people of an idea whose time has come. A direct analogy, to this whole discussion of digital sound recording, is the appearance in the late-Twenties of the microphone boom. In the early days of sound films large condenser (really!) mikes were buried in flower pots and hung with ropes over the sets. It was only a matter of time before directors and actors realized that it was unnatural and downright stupid to have to play to the microphone, and so the moving microphone was born. While there are many accounts of who *really* invented the microphone boom, this situation clearly falls under the category of "parallel invention."

It's almost embarrasing watching adults rush to be a footnote in the history of film sound as "the first right-handed Taurus to mix an all-digital sitcom." I mean, it's one thing to design and build a piece of equipment and use it in a novel manner; its quite another to take a digital recorder off the shelf and play "go fish" for compliments.

Sometimes the hype is so extravagant, that it's funny: In 1983, two seminal mid-Sixties rock films, *The T.A.M.I. Show* and *The Big TNT Show* were re-issued as one film titled *That Was Rock*. When the film came to Los Angeles, the ads proudly proclaimed "In New Digital Stereo." Say what?

If my understanding is correct, these films were originally shot on two-inch videotape, and were most probably recorded in mono. I base this assumption partly on the fact that Phil Spector produced *The T.A.M.I. Show*. In any event, what I heard at the Vogue Theater on Hollywood Boulevard (one of the few theaters in the Los Angeles area so-equipped to show the film in "digital stereo," according to the ad) was distorted beyond all reason. No mono film since the development of optical "ground noise" reduction in the early-Thirties has sounded this bad.

Undoubtedly some poor soul, having just bought a CD player, paid five bucks expecting to hear that kind of sound quality in a movie theater. Okay, I'll admit that this example is a bit extreme. But you get the point.

It seems that the limiting factor in sound quality is that intangible called "taste": with music, where the mikes are placed and the feel of the arrangement and performance. In film

tember 1953.

Since that time, double-system projection has rarely been used in standard theatrical motion pictures, although it sees daily use for special formats such as IMAX/OMNIMAX and Showscan, both of which employ interlocked 35mm six-track reproducers.

There seems to be an informal, but near-unanimous consensus among Hollywood film-sound people that the digital exhibition format of the future should have eight tracks. The most often mentioned format would have five full-frequency channels behind the screen (as in dozens of 70mm "discrete" six-track films released B.D. — Before Dolby), separate left and right surrounds, and a subwoofer track. Such a format would be all things to everybody, whereas today, you can have discrete six-track or split surrounds but not both. The addition of two more tracks will give mixers — experienced in six-track Dolby Stereo, a flexible tool.

Digital Fluorescentsound

There is a widely held belief in Hollywood that single system — sound carried on the picture print — is the *only* way to go. Double-system interlock projection poses the ever-present problem of sync error, either caused

MISREPRESENTATION — continued . . .

sound, it's the choice of the sound effects, where they are placed, and how the combination of dialog, music and effects works with the image. It doesn't do any good if a dialog track is "all digital" if it sounds like an Italian Western; albeit a clean, all-digital Italian Western.

One can see the 1933 version of *King Kong*, and still marvel at the incredible sound effects by Murray Spivack, working with just three playback dubbers! And what about Ben Burtt's sound effects for the original *Star Wars* film, using a four-track Tascam deck.

The freshness and high quality of the above work has not yet been dimmed by time, and probably will never be. Possibly to overstate the point, all of the above are remembered not because they used "state-of-the-art" equipment — in fact, they didn't. Instead, they will be remembered because they advanced the state of the art and craft of film sound itself.

This writer is not ignorant of the necessity for new, expensive toys to "earn" their keep. The people who sign checks in this town don't care about anti-aliasing filters: they want to know what effect the expensive digital multitrack is going to have on *their* bottom line.

It has to be clear that this commentary does not intend to slight or downgrade the efforts that have been undertaken by the few studios that have used (and promoted the use of) digital recording for films. Whatever one might say about where their mouths are, they are putting hard-earned cash there. And, when the big leap is taken to random-access digital sound for films, expect to see these same studios lap their competitors before they get out of the starting blocks. Translation: In this instance, recording by the numbers *will* mean revenue — for the sound department, that is, not at the box office.

This author is neither a Luddite nor a Digiphobe. Perhaps my beef is that I believe the primary impact digital recording will have on films will be felt primarily by the filmmakers. I am thinking here most specifically of random-access picture and sound editing and, as has been said many times, these tools will do for filmmakers what personal computers have done for writers.

But as long as we think it's stupid for a writers to cite the use of a dedicated word processor and a laser printer as proof positive of literary quality, let's *not* try to pass the presence of digital recorders as endowing a track with taste.

The perceived difference in sound quality will be unnoticed by 95% of the audience. Which is quite fine with me. Films made today using digital recordings sound almost identical to high-quality standard analog mag films. Even if the difference *is* noticeable, why would we want to wave a red flag at the public by telling them to listen to the low wow and flutter, when all they want to do is laugh and cry? □□□

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by a malfunctioning synchronization device, or simply with the sound for reel 5AB being played with reel 6AB!

The anticipated leap to digital reproduction in motion picture theaters has brought the single/double system controversy to the fore again. The obvious places for the soundtrack — magnetic stripes on 35mm or 70mm prints, and an optical track on 35mm print — do not appear to have sufficient bandwidth to handle multichannel digital audio soundtracks with professional standards of 16-bit resolution and a 48-kHz sampling frequency.

The best-known single-system idea is that proposed by Peter Custer in his Digital Fluorescentsound process. Eight, 16-bit/48-kHz channels of digital audio are "recorded as colorless and transparent, brightly fluorescent high-density data image, multiplexed over the picture across the entire photographic image space."

While the process currently exists only in the form of patents, Custer hopes to obtain financing within the industry to create a prototype system. He anticipates that the Fluorescentsound cinema processors will be leased to theaters, both to sidestep the high cost outlay by the theater owners, and also to assure proper maintenance.

Because the Digital Fluorescentsound "soundtrack" is invisible, standard Dolby Stereo optical soundtracks will provide compatibility with unconverted theaters. Another important benefit of the system is that it would provide top quality sound without the cost and trouble of preparing 70mm prints (approximately $12,000 each) which, in turn, costs the major studios millions of dollars each year.

Double-System Digital Projection

If someone attempts to sell a double-system digital interlock format to the industry, the first question people ask is something like "Can it be operated reliably and without worry in the Spearfish, South Dakota Cinema XIV?"

Discussed below are two custom interlock systems that have been used to showcase the ways in which all-digital soundtracks can benefit from digital presentation; neither system is being proposed for adoption by the industry.

Early this year, Walt Disney Productions once again scored a first with *Fantasia*, this time with the first public presentation of that landmark stereo film's digital re-issue in digital interlock. Special equipment was
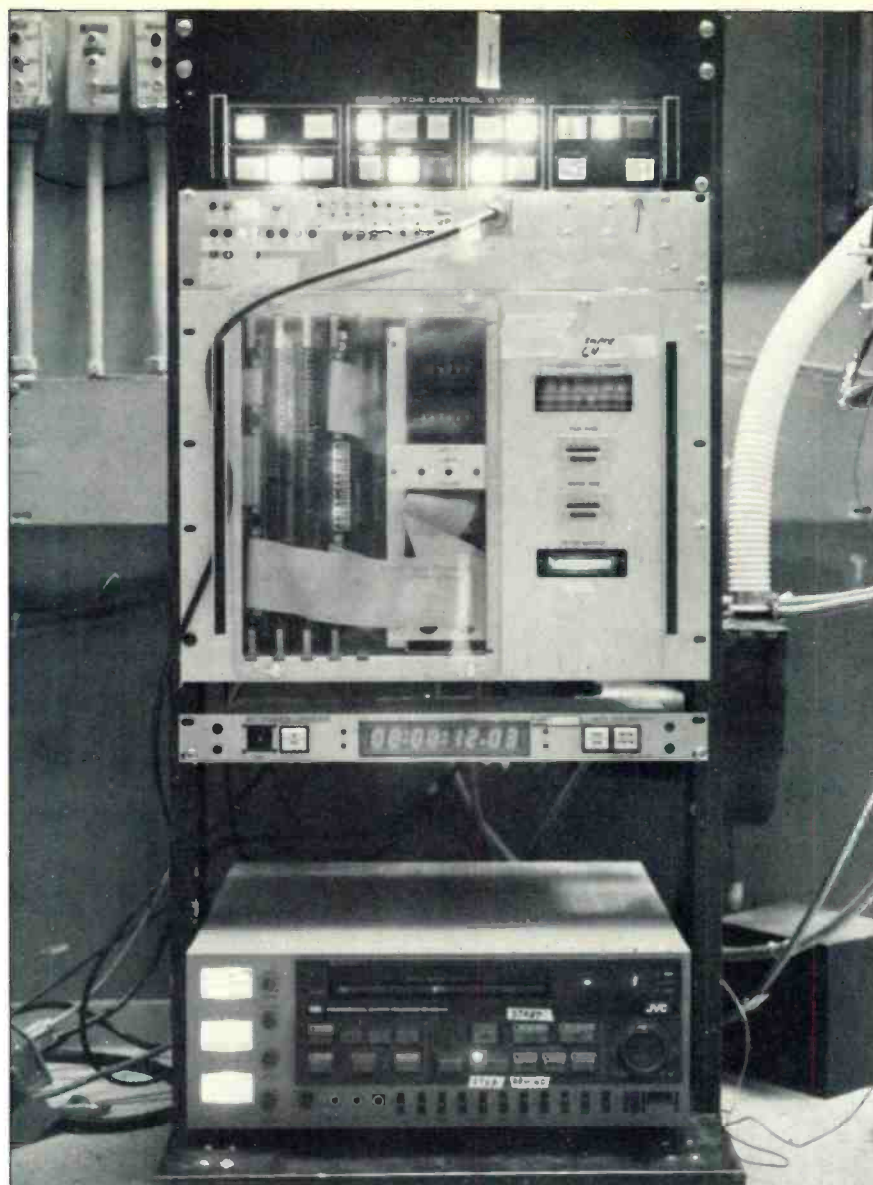
A custom Disney projector controller unit used to provide digital interlock with an Audio+Design/Calrec-modified Sony PCM-710 processor for double-system presentations of the classic movie *Fantasia*.

installed at the Plitt's Century Plaza Theater II in Los Angeles, beginning February 8, 1985. The digital Lt-Rt master was transferred in a continuous segment onto a Sony PCM-F1-encoded one-inch videotape which, in turn, was copied onto a F1-encoded half-inch VHS cassette to provide the necessary two-hour running time. The same digital program has since been presented at the Avalon Theater in Washington D.C., and the Ziegfeld Theater in New York City.

The system utilized a proprietary projector drive system with stepping motors developed for use in the EPCOT Center film shows to interlock projectors with analog multi-tracks; all film shows at Disney theme parks utilize double-system projection. The PCM-encoded tape was played through with an Audio+Design-modified Sony 701ES processor, with SMPTE timecode on linear tracks of

the videocassette loaded in the JVC-BR-8600 industrial VCR.

The projector drive system always knows "where" the 35mm print of *Fantasia* was, since it is aligned at the 12-foot Academy Picture Start frame. The 24-frame drop-frame timecode on the videocassette was recorded separately from the recording of the PCM-encoded digital audio, with the timecode starting approximately two mintues before the first frame of picture, and the numbers bearing no particular relationship to the picture.

At the Century Plaza, it was determined that the projector motor had to be started 64 seconds after the cassette. Any variance caused by starting the projector sooner or later is compensated by the projector-drive mechanism, which regards the 64-second start mark as the nominal sound start mark, and knows the correct picture frame that has to be in the

gate at a given moment. Thus, two hours into the film, the projector controller knows what SMPTE timecode number matches frame #172,800.

Of course, since there are no timecode numbers on the film, if a film break exists on the print, the system would not be able to re-sync. This arrangement was admittedly a "one-off" system designed to show the industry that the hardware for digital reproduction in theaters is available today.

In August 1984, Giorgio Moroder's all-digital re-issue of Fritz Lang's *Metropolis* had a one-time-only digital showing at the Samuel Goldwyn Theater of the Academy of Motion Picture Arts and Sciences utilizing two PCM-3324 digital multitracks interlocked to the projectors with the standard practice of SMPTE timecode on a dubber. Two multitracks were needed because of projector changeovers.

Showscan

One of the first uses of digital sound-on-disc (shades of Vitaphone!) projected in interlock began in Toronto this past November for the Showscan film *Tour of the Universe*, produced for Interactive Entertainment, Inc. [Showscan, which uses 70mm film photographed and projected at 60 fps, was described by this author in the April 1984 issue of *R-e/p* — Editor.]

A Constant Angular Velocity (CAV) LaserVision disk containing two digital and two FM-encoded tracks is interlocked to the custom Showscan electronic projector. According to Showscan sound engineer John Ruck, "the subwoofer track is encoded onto one of the digital channels, and is separated by an active crossover in playback." The design requirements for the 18 by 20 foot space shuttle simulator, which was the "theater" for *Tour of the Universe*, called for four full-frequency channels (left and right channels for front and rear), with no center channel.

For additional channels in standard large (90 by 82 feet) Showscan theaters, the company is investigating two all-digital six-channel formats: three Compact Discs running in interlock, and optical digital tracks recorded on a composite 70mm print. (Prior to *Tour of the Universe*, Showscan sound came from interlocked Dolby-encoded 24 fps 35mm mag.)

The Compact Disc format will be first seen for two films at the Vancouver Expo '86 beginning in May. Three industrial, customized CD players will be cascaded for six-track reproduction. The "bullet-proofing" has entailed going into the unused user bits of the PQ Subcode, not only

for synchronization, but also to ensure that disc #1, and not Yes' *90125*, is inserted into player #1. An incorrect disc will be rejected like a crumpled bill from a coin changer.

The optical format, which is being investigated in conjunction with Eastman Kodak, will take advantage of the very high speed (56.1 ips) of the 60 fps Showscan format to allow sufficient bandwidth to place digital tracks where there are magnetic stripes on standard 24 fps 70mm prints. Whether the system will be applicable to standard 24 fps 35mm and 70mm projection remains to be seen.

Current plans for the Showscan

Film Corporation are to build 30 digitally equipped theaters around the world by the end of 1986.

The major players in the digital film sound field: Sony, Mitsubishi, The Droid Works and CompuSonics, have made no announcement yet regarding any forthcoming systems, although all have expressed interest in bringing digital sound to theaters. The biggest force in theater cinema processors, Dolby Laboratories, has also made no announcement of its plans, if any, to throw its hat in the ring; one obvious possibility is the use of the ADM (Adaptive Delta Modulation) technology that it has developed for satellite broadcasts. ∎∎∎

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Northeast:

☐ **GIANT SOUND** (New York City) has opened for business as a 24-track studio. Featuring a 32-input **Trident 80B** console, plus an **Otari MTR-90** multitrack, **MTR-12** half-inch, and **MTR-12** quarter-inch machines, the new facility is slated as a "full service, audio studio," according to chief engineer **Joe Salvato**. Outboard gear includes a **Quantec Room Simulator**, an **AMS DMX 15-80S** DDL/pitch shifter, four **Valley People DynaMite** compressor limiters, a **dbx Model 900** rack with noise gates, an **Orban** stereo parametric equalizer, and an **Aphex II Aural Exciter**. The microphone collection is comprised of **AKG C12s**, **Neumann M47s**, and supplemental models from **Sennheiser**, **Shure** and **Electro-Voice**. Studio monitoring is provided by **Meyer 833s**, **Yamaha NS-10Ms**, and **E-V Sentry 100s**. *1776 Broadway, New York, NY 10019. (212) 247-1160.*

☐ **WESTRAX** (New York City) has expanded its electronic-music capabilities with the purchase of an **IBM PC** and sequencer software from **Octave Plateau**. The facility's synthesizer array consists of **Yamaha DX-7** and **TX-216** rack, a **Korg Poly-800**, and a **Casio CZ101**. In addition, the **Sound Workshop Series 30** board has been expanded to 36 inputs, and an **Otari MTR-12** half-inch two-track mastering machine added to the equipment list. Outboard equipment acquisitions include a **Lexicon PCM-42** digital delay, a **Studio Technologies Ecoplate III** reverb unit, and a **Korg SD-2000** sampler/digital delay. *Manhattan Plaza, Basement Level, 484 West 43rd Street, New York NY 10036. (212) 947-0533.*

☐ **UNIQUE RECORDING** (New York City) has re-opened Studio A after extensive renovation. The control room now boasts a 48-input **Solid State Logic 4000E** console with **Total Recall**, linked to twin **Otari MTR-90 MkII** 24-tracks, and **Studer A80** half-inch two-track machines. **Sony BVU-800** three-quarter inch and **Beta/VHS Hi-Fi** video decks are also available. The room's acoustics were designed by co-owner **Bobby Nathan**, chief of maintenance **Bruce Freeman**, and **Al Fierstein** of **Acoustilog**. "With the [Total] Recall feature of the SSL, keeping track of large MIDI set-ups is a breeze," says Nathan. In addition, **MIDI City**, a MIDI-based studio within the facility, has obtained a vintage 32-channel **Neve 8068 Mark II** console containing eight VCA subgroups, and the monitoring system upgraded to include **UREI 813s**. *701 Seventh Street, New York, NY 10036. (212) 921-1711.*

☐ **HILLSIDE SOUND STUDIO** (Engelwood, New Jersey) is a new 24-track facility aimed at album production, audio-for-video post production, and electronic-music recording. The control room features a 24-by-24 **Troisi** mixing console, a **Studer A80 Mk IV** 24-track, **A80** half-track and **A810** half-track with center-track SMPTE time-code. Outboard gear for the new complex includes a **Lexicon 224XL** digital reverb and **Prime Time II** effects processor, **Eventide Harmonizer** and **Flanger**, **Valley People Kepex** noise gates and **Gain Brains**, and "full MIDI-interfacing capabilities." In-house instruments comprise a **Linn Drum**, a **Simmons** electronic drum system, **Yamaha DX-7**, **ARP 2600**, **Oberheim** synthesizer, and an **E-mu Systems Emulator II**. A full complement of microphones from **Neumann**, **Electro-Voice**, **AKG**, and **Shure** round off the equipment list. Staff engineers for the facility are **Dae Bennett**, **Dave Kowalski**, and **Paul Mufson**. *102 Hillside Avenue, Engelwood, NJ 07631. (201) 568-3268.*

HILLSIDE — new 24-track facility

☐ **STUDIOLINE CABLE STEREO** (Reston, Vermont) has purchased 48 **Studer A810** two-track tape machines for use in the company's main production/origination division. The machines will be used for production of program-master tapes, as well as for direct playback into the system. *11490 Commerce Park Drive, Reston, VA 22091. (703) 648-3200.*

☐ **UCA RECORDING** (Utica, New York) has added two **Telefunken U-47** tube microphones and a **Neumann U-87** to its microphone collection. Other new equipment includes an **Oberheim DXa** drum machine with MIDI capabilities, and an "extensive sound library." *1310 Lenox Avenue, Utica, NY 13502. (315) 733-7237.*

☐ **POWER PLAY STUDIOS** (Long Island, New York) has purchased **Yamaha REV-1** and **REV-7** digital reverbs, a **Roland DDR 30** digital drum rack, two **Pultec** equalizers, a **Publison Infernal Machine** digtal reverb/sampler, an **E-mu Systems Emulator II** digital synthesizer. *38-12 30th Street, LIC, 11101. (212) 729-1780.*

☐ **C/M STUDIOS** (New York City) is a new 48-track commercial music and album production facility, that specializes in electronic tracking. The new complex, comprising Studios A and B, went on-line in December 1985. Studio A boasts a 37-input **Amek Angela** console with **Audio Kinetics Master Mix** VCA automation linked to **Otari MTR-90 MkI** and **MTR-90 Mk II** 24-tracks, and an **MTR 12** half-inch two-track. Accompanying outboard gear comprises a **Yamaha REV-1** digital reverb and **DDL 1500** effects processor, **Drawmer** noise gates, an **Eventide SP-2016** digital effects unit, an **URSA MAJOR MSP-126**, **Lexicon 224XL** and **Model 200** digital reverbs. Chief engineer **Brian Lee** says facility's custom-designed MIDI-multiplex system, which is hard-wired under the control-room floor to corresponding keyboards. In-house instruments consist of a **NED Synclavier II** digital synthesizer, two **Yamaha TX-16** racks and **KX88** keyboard controller; a **Roland Super Jupiter** and **CBX** synthesizer; **Sequential Circuits Prophet** and

C/M — featuring Amek Angela console

T-8 and a **Linn 9000 MIDI** sequencer. The above keyboards are interconnected via **Apple IIe** and **Commodore 64** personal computers, running appropriate MIDI-based sequencing software. **Studio B** features a 24-input **Teac Model 15** console, and an **Otari MTR-90** 24-track machine. This smaller studio is used mostly for pre-production projects, for album or commercial scores. *30 East 23rd Street, New York, NY. (212) 777-7755.*

Midwest:

☐ **GNOME SOUND** (Detroit, Michigan) has added to its exisiting MIDI capability with the acquisition of a **Lexicon PCM-70** MIDI-capable digital processor. In addition, the studio recently completed mastering to a **Mitsubishi X-80** digital two-track for guitarist Bobby Barth. *9918 Lauder, Detroit, MI 48227 (313) 835-0169.*

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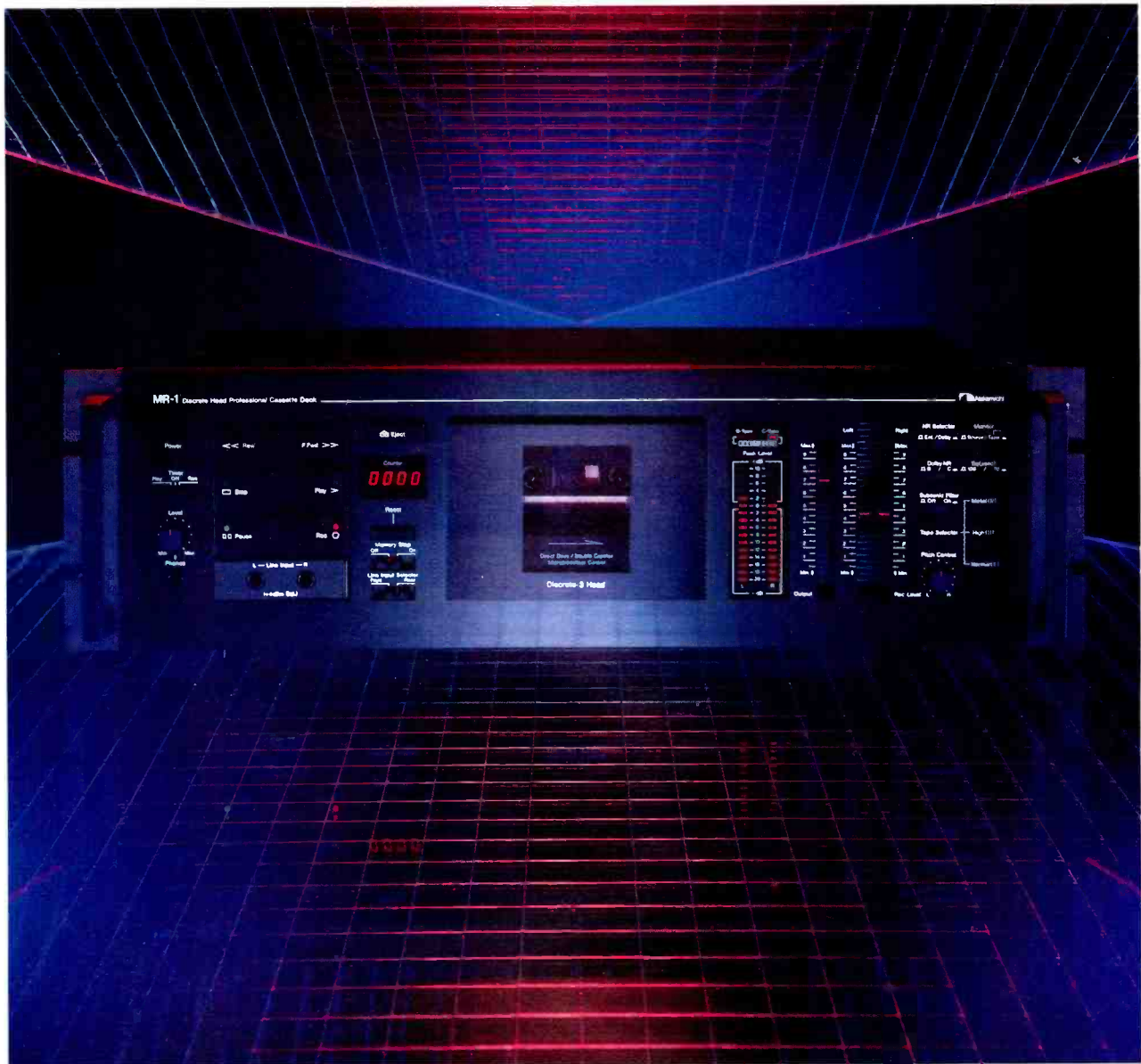
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□ **SOLID SOUND** (Ann Arbor, Michigan) is a 24-track facility that recently added audio-for-video synchronizing equipment, including an **Alpha/Sony 5850** VTR, and **Adams-Smith 2600** timecode synchronizer. Additional hardware includes **Sequential Circuits** 23-bit sampling keyboard/synthesizer, **NEC** video monitors, an **Otari MTR-12** four-track mastering machine, and an **Aphex Compellor** limiter. *P.O. Box 7611 Ann Arbor, MI 48107 (313) 662-0667.*

□ **SWEET SOUNDS** (Cleveland, Ohio) has opened its new 16-track production facility designed by **Pi Keyboards and Audio**. The control room centers around a 24-input **Soundcraft Model 500** console linked to an **Otari MX-5050 MkII** eight-track machine with remote control and autolocator, an **MX-5050 MkIII** four-track, and an **MX5050 MkIII** two-track. Outboard gear includes a **Lexicon PCM-60** and **PCM-42**, an **Eventide H910 Harmonizer**, a **Symetrics 522** combined compressor, expander, limiter, gate, ducker, and an **Aphex Type B Aural Exciter**. Monitoring is provided by **JBL Model 4312s, 4401s**, and **Aurotones** — all of which are driven by **Crown DC300A** Series II amplifiers. In-house instruments for the new facility include an **E-mu Systems Emulator II**, an **Ensoniq Mirage**, an **Oberheim OB-8, DSX** and **DMX**, a **Yamaha DX-7**, **Roland TR-707** and **TR-727** drum computers, a **360 Systems MIDI Bass**, and a **Yamaha U1J** piano. Audio-for-video synchronization is provided by a **Synchronous Technologies SMPL System**, and an **Apple IIe** personal computer. According the facility's president, **Allen J. Friedman**, Sweet Sounds' production projects will be aimed at film and television scoring, audio-visual multi-image soundtracks, jingle production, scoring complete Music Videos, and demo projects for local artists. *4098 Washington Blvd., Cleveland, OH 44118. (216) 292-0787.*

SWEET SOUND — new 16-track studio

Southeast:

□ **ALPHA AUDIO RECORDING** (Richmond, Virginia) claims to be the first studio in the Southeast region to add a Compact-Disc version of the new **Dewolfe Music Library** to its existing music library. This acquisition is coupled with the studio's recent purchase of the **Sound Ideas Sound Effects Library**, which is also on Compact Disc. *2049 West Broad Street, Richmand, VA 23220. (804) 358-3852.*

□ **CRAWFORD POST PRODUCTIONS** (Atlanta, Georgia) has purchased three **Graham-Patten Systems Model 612 EditSuite** mixers for video post production. The units are said to contain direct serial interface capabilities for a variety of video editors, and a programmable EQ. When operating with an editor, the integrated system can control audio source selection, audio preview, and transition rates and starts. *535 Plasimour Drive, Atlanta, GA 30324. (404) 876-7149.*

☐ **MEGA SOUND** (Bailey, North Carolina) has added an automated 32-input **Harrsion** and **Autoset** automation system to the facility, along with a **Lexicon 224XL** digital reverb with **LARC**, a **Linn Drum**, an **AMS DMX 15-80S** digital delay, the **Yamaha REV-7** digital reverb. Monitors are **Yamaha NS-10Ms**. *P.O. Box 189, Bailey, NC. (919) 235-3362.*

☐ **STRAWBERRY JAMM** (West Columbia, South Carolina) has taken delivery of a new **Sony/MCI JH-110 C** two-track machine. This acquisition is said to complement the studio's fully-loaded **Sony/MCI JH-636** automated console and **JH-24** multitrack. *3964 Apian Way, West Columbia, SC 29169 (803) 359-4540.*

Mountain:

☐ **Colorado Sound** (Westminster, Colorado) has purchased an **Otari** MTR-90 24-track, a **Lexicon PCM-70** MIDI-capable digital processor, and has modified the facility's exisiting **E-mu Systems Emulator II**, to enable the keyboard to generate SMPTE timecode for audio-for-video sychronization. *3100 West 71st Avenue, Westminster, CO 80030. (303) 430-8811.*

Southern California:

☐ **UNIVERSAL CITY STUDIOS** (Universal City) has taken delivery of what is claimed to be the largest **Harrison PP-1** stereo post-production console equipped with hard-disk automation. Installed in the studio's Dubbing Two theatre, the board is slated for use during stereo post-production for network dramatic productions. The console features dynamic, timecode referenced automation of 13 functions on each of the system's 81 input and 24 submasters. *100 Universal City Plaza, Universal City, CA 91368. (818) 985-4321.*

☐ **LE MOBILE** (Encino) has appointed **Abe Hoch** as vice-president of Le Mobile Inc., according to company president **Guy Charbonneau**, who says that this new position illustrates "the unique bi-coastal capabilities of Le Mobile." *PO Box 1842, Suite 790 Encino, CA 91426. (818) 992-8481.*

☐ **EVERGREEN RECORDING** (Burbank) has acquired a 54-input, six-channel film scoring version of the **Harrison MR-2** automated production console. The purchase is said to be part of the **CBS/MTM Scoring Stage** renovation. The console will feature "split" three-channel panning, isolated 48-channel multitrack output routing, and two-channel isolated stereo output for stereo mixdown from 48-channel multitrack or six-track film. *4403 West Magnolia, Burbank, CA 91505 (818) 841-6800.*

☐ **VOICE OVER L.A.** (Hollywood), a recently opened eight-track facility geared for radio, television, and film-sound productions, has purchased a **Sound Ideas Compact Disc** sound effects library with over 3,000 effects, and **Audio Kinetics Eclipse 410** synchronizer/editor with list management display for audio-for-video, using SMPTE timecode interfaces. *1717 North Highland Avenue, Hollywood, CA (213) 463-8652.*

☐ **IMAGE RECORDING** (Hollywood) has added a new 24-track studio to its existing 48-track facility. Studio B is a limited partnership between Image Recording principals **Harry Maslin** and **John Van Ness**, and **Redwing Studio** owner **Tom Seufert**.

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Constructed from a former mastering studio, Studio B — aimed at audio-for-video, film scoring and album projects — has been named a "MIDI/SMPTE interlock room," and emphasizes electronic instrument tracking for Stereo TV and film scoring. The 24-by-13-foot control room is centered around MIDI-capable equipment, which includes: a **Linn 9000** drum machine/MIDI recorder with digitally-sampling capabilities and 3.5 microfloppy disk drive; the **Yamaha YCAM system**, consisting of a **TX816** rack, a **DX-7**, and a **KX88** — a MIDI-capable **Rhodes Chroma**; an **E-mu Systems Emulator II**; a **Rhodes Juno 106**; 24 MIDI-In jacks patched to the console; plus MIDI-jack and mike panels located on each wall. All of the above equipment is interconnected by a 1024K **Apple Macintosh** computer running **Southworth Music Systems' Total Music** sequence software. For mixing and recording, a **Trident Series B** 56-input console linked to a transformerless **Sony/MCI JH-24** and **Ampex ATR-102** half-inch two-track is complemented by an array of outboard gear, comprising a **Yamaha REV-7** digital reverb, **EMT** plate reverb, **Lexcion 200** processor, **dbx Model 165** noise reduction; and monitoring provided by UREI 813s and **Yamaha NS-10Ms**. Synchronization is provided by **Timeline** timecode modules and a **Roland Sync Box 80**. A collection of microphones from **Neumann** and **Shure** complete the recording equipment list. Video equipment consists of a **JVC 6650** three-quarter inch VCR and **Proton** 25-inch monitor. Adjacent to the control room is a 400 square foot "L"-shaped isolation booth used for vocals, ADR, Foley, and for drum and piano isolation. *1020 North Sycamore, Los Angeles, CA. 90038 (213) 850-1030.*

Foreign:

□ **SSVC SERVICES** (England) has ordered 16 **Neve 5322** "on-air" stereo broadcast consoles for distribution to its military broadcast stations in Hong Kong, The Falklands, Cyprus, Gibraltar, Germany, and other countries where British service men reside. The desks feature stereo input modlues which accept input from cart machines, tape recorders, and other line level sources. *London, England.*

□ **WESSEX STUDIOS** (London, England) has acquired a **Mitsubishi X-850** digital 32-track equipped with razor-blade editing capabilities. This purchase will complement the facilty's exisiting SSL-equipped Studio One. *London, England. England.*

□ **TAPE ONE** (London, England), reportedly the world's first all-digital CD mastering facility, has acquired a second **Neve DSP** console. The board was installed in a new **Tom Hidley**-designed room, which also features a **Neumann VMS-80** disk-cutting lathe, **Sony PCM-1610** and **PCM-701** digital processor, a **Mitsubishi X-80** two-track digital mastering machine, and a **Studer A820** analog deck. *29/30 Windmill Street, London, England WP 1HP.*

□ **PHILIPPE SARDE'S STUDIO** (Paris, France) has purchased a 60-input **Neve** console for use on film productions. The console features **NECAM 96** fader automation, FSE equalizers, 48-track routing, 48 "bargraph"-type meters, and a comprehesive solo system. *Paris, France.*

STUDIO PROFILE: Blue Jay Re-opens as a 46-track Facility

☐ **BLUE JAY RECORDING** (Carlisle, Massachusetts) has re-opened after major renovations, and now describes itself as the first 48-track **Solid State Logic** facility in the New England area. The redesigned control room houses a 48-input **SSL4000E** with **Primary Computer, Total Recall,** and reportedly one of the largest patchbays SSL has ever produced. In the adjacent machine room, new transport acquisitions include a **Studer A-800 Mk III, A80** two-track with half-inch heads, **A820** with center-track timecode, and a **dbx Model 700 CDPM** digital processor. **Lynx TimeLine** modules are used to synchronize all tape transports. Outboards include a **Lexicon** units **224XL, PCM-60,** and two **PCM-42s**; **Drawmer** noise gates, plus **AMS RMX-16** reverb and **DMX-15-80S** digital delay and sampling unit.

Noise reduction is provided by **Dolby** Type A M-16 and Type A 361 units, and a **dbx Model 180.** In-house musical instrument encompass a **Kurzweil 250** with 50 Kbytes of RAM, and an "extensive library of sounds" stored on the **Macintosh** computer. The studio and control room were redesigned by **Russell Berger** of the **Joiner-Rose Group,** who employed the "largest RPG diffusor system to date. The result is a decay time that is very even over a broad range of frequencies," says owner **Bob Lawson.** In addition, the studio utilizes the "world's only glass RPG diffusor, which doubles as a picture window." Lawson speculates that the SSL acquisition will move his studio into a position to better

Blue Jay — recent renovations including SSL 4000E

accommodate the studio's existing major-label album projects. The studio is also "moving into the audio-for-video and audio-for-film direction," he adds. "With the computer capabilities on the SSL, it is easy to compile and mix many sources [music, dialog, and sound effects] in a manageable and efficient way." *669 Bedford Road, Carlisle, MA 01741. (617) 369-2200.*

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THE SCANDINAVIAN CONNECTION:
Recording Miles Davis' *Aura* at Easy Sound

by David Rideau

When Miles Davis goes into the studio, it's always news. But when he travels to Europe to record with a large ensemble, it instantly becomes an event. In this case a report on the session also makes for interesting reading (I hope!), if only to see how a relatively unknown Danish studio handled a project of this magnitude.

Miles Davis should not need a formal introduction; his music has not only influenced jazz, as well as other musical forms, it has almost charted its course. But this is not an article on "The Man," which would take more space than I'm allowed — our concern here is with the recording session.

Easy Sound Recording, Copenhagen, Denmark, is a very special place to me, probably because I helped build it! That fact aside, the facility is basically a converted film theatre with over 5,000 square feet of recording area, including three isolation rooms and a large stage with a ceiling height of 30 feet. [An article describing the design and construction of Easy Sound appeared in the October 1982 issue of *R-e/p — Editor.*]

Easy Sound is a popular room in northern Europe for recording acoustical instruments ranging from string quartets to "live" drums for rock and roll sessions. At its opening, the room was equipped with a small API con-

sole that has since been moved to the recently constructed "B" studio located in the basement, which has 1,400 square feet of recording area. The API in Studio A was replaced with a 40-input frame Solid State Logic SL4000E console, currently stocked with 34 input channels, and the most recent computer package offered for automated mixing. The studio is also equipped with a Studer A-800 24-track, plus Studer A-80 and Sony MCI JH-110 two-tracks. Also available is a Sony PCM-1610 processor for digital mastering. As far as outboard gear goes, Easy Sound has more than its share of quality "toys" needed in the modern recording world. Last, but not least, is the studio's collection of 60 or so microphones that would make most stateside studio owners green with envy: for example, the facility boasts six Neumann "tube-type" U-47s.

The facility is owned and operated by two brothers, Niels-Erik and Henrik Lund who, at the tender ages of 25 and 27, have already been studio owners for the last 10 years.

Birth of a Session
It all began when Miles Davis received last year's Sonnings Music Award, a Danish prize presented to persons of outstanding musical merit; in the past the SMA has been awarded to Igor Stravinsky, and many other international musical legends. As a

ROOM LAYOUT FOR *AURA* SESSIONS

MILES DAVIS IN DENMARK

part of the festivities, Palle Mikkelborg, a Danish musician-composer, was commissioned to write a piece, titled *Aura*, that would be performed on awards night, with Miles Davis as soloist. Davis was so impressed by his colleague's instrumental work that he decided to return to Copenhagen to record *Aura* in preparation for a later album release. It was intended that the same ensemble record the piece, with the addition of several local and international players.

When the initial mental count of musicians and required console inputs was done by the brothers Lund, there was a rude awakening: there would not be nearly enough inputs! Two alternatives were considered: give up the booking of a lifetime; or get more inputs. Without hesitation they chose the latter. The plan was relatively

simple on paper — bring up the trusty old API from Studio B downstairs! They knew it would involve a great deal of work to devise an appropriate interconnect scheme to enable the setup to function in a sane, professional manner. Not to mention that the interconnect cabling had to be done fast to avoid downtime in the two rooms involved.

It was also becoming more and more clear that 24 tape tracks would not be enough, so a second 24-track from the basement "rose" to the occasion (sorry). Setup time for 46-track was minimal, since the studio owned a timecode synchronizer, and had done many multiple-machine sessions in the past. Studio staff took the time to pre-stripe the blank tape reels with

Microphone List
Drums: rack toms AKG C414s; floor toms C414s; cymbals B&Ks; bass drum Electro-Voice RE-20; snare Shure SM56; high-hat AKG C451 with CK-1 capsule.
Acoustic piano: Neumann U87 low; C414 high.
Assorted percussion: four AKG C451s.
Congas/bongos: C451.
Trumpets: RE-20s.
Trombones: Neumann U-47 (tube-type).
Saxophones/flute: three Telefunken M-49s and two C414s.
Harp: Direct and B&K.
Guitar amps: C414.
Acoustic bass: Neumann SM2 (tube-type) combined with his own stereo pickup.
Electric bass: direct out from effects rack; RE-20 on cabinet.
Miles: U-47 (tube-type) with windscreen.

Close-up detail of drum and percussion miking at Easy Sound.

MILES DAVIS IN DENMARK

SMPTE code, thereby making sure that there were no sync problems later.

During the same time period (two days) Miles arrived in Copenhagen, and proceeded to meet with Palle and go over the score to make any last-minute changes.

Studio Setup

As can be seen from the accompanying room and mike layout diagram, the idea was to have 27 musicians plus Miles set up in a semi-isolated but sound enhancing environment, which would be sufficiently versatile for live or overdub recording. They also had the players placed so that they were close enough to perform without headphones, if they so desired.

The drums were set up in the middle of the studio, loosely surrounded by four-foot gobos to enable eye contact with the conductor and other musicians. An overhead semi-absorbent parasol could be raised or lowered to acoustic taste. On the drummer's immediate right was a full Simmons electronic drum set which, acoustically, was not a problem but surely didn't help the microphone input situation.

To the left of the drums were the acoustic and electric basses, and concert grand piano. Behind the drummer were two percussionists who, together, seemed to have just about every percussive instrument ever made.

Palle, being conductor, stood directly in front of the drummer in a highly visible position. To his left were the horns: five trombones, five trumpets, and five saxophones placed in a semi-circle. A tight line of gobos divided this section from the rest of the group. To the conductor's right were two electronic keyboardists, and the guitarists.

Behind the conductor was Miles, who had two tall gobos placed directly behind him for minimal acoustical isolation from the rest of the players. The idea of this setup was to provide maximum visibility, with the gobos helping to arrest a large part of the leakage coming in on the "live" side of the microphone, which was set to a cardioid pattern. (The same principle was used with the horn section, by placing gobos loosely behind the players.)

Recording Sessions

There were to be only five days of recording, with Miles leaving on the sixth day; they *had* to stay within the production schedule. (Keep in mind that, at this point in the proceedings, the participants were still in the dark as far as who would play what, when

Miles Davis (*left*) and session conductor/producer/composer Palle Mikkelborg.

and, as a result, everyone wanted to be prepared for anything.)

The engineering crew consisted of the Lund brothers as co-head engineers; the two consoles kept their hands full throughout the session. Second engineering and maintenance tasks were handled by Niels-Erik Otto and Henrik Jensen.

The two first days of the session proved to be a gradual warm-up process, for which everyone was present on the floor except the horns. As Miles and the rhythm section ran through the charts, he went around to each player commenting on what he did or didn't like, and made small arrangement changes as the piece progressed. These small suggestions, or maybe the method itself, created a positive

ambience in which the musicians could generate their best performance. The best moments of these days were recorded onto 10, two-inch reels of tape, and this was just the beginning!

Saturday turned out to be the ultimate test. The basic ensemble and horns showed up in the morning, and the boys worked fast, recording 10 more reels of two-inch before the early afternoon. I can't fathom how this session, which seemed like total chaos on the surface, rolled like clockwork in real-time. To add to the confusion, a six-man film crew from Danish TV was filming the event for a special broadcast. Now the photographers and journalists — they were literally everywhere! (Keep in mind that to the six million people of Denmark, this was the *biggest* recording session since Leonard Bernstein recorded Carl Nielsen in the composer's homeland.)

The five-day session turned a normally spacious and comfortable studio environment into a crowded beehive of activity. At one time the control

Session Musicians

Miles Davis — trumpet.
Palle Mikkelborg — producer, composer, conductor, trumpet player.
John McLaughlin — guitar.
Vince Wilburn — Simmons electronic drums.
Lennart Gruvstedt — acoustic drums.
Neils Henning Orsted Pedersen — acoustic bass.
Bo Stief — electric bass.
Ethan Weisgard — congas, bongos, timbales.
Marilyn Mazur — percussion.
Bjarne Roupe — guitar.
Thomas Clausen — keyboards.
Kenneth Knudsen — keyboards.
Ole Kock Hansen — acoustic piano.
Danish Radio Big Band.

room head count reached 25 people —a situation compounded by the fact that there was a second console and 24-track in the same area, making for some tight squeezes.

At 2 p.m. the studio was cleared for a special guest artist to record some overdubs: John McLaughlin popped into the studio, smiling as though this was his first session as studio guitarist. Since McLaughlin was guitarless, Mikkel Nordso, a local player with a happening quiver of guitars, came up with a Gibson to his liking.

Sunday began with the complete ensemble again, and graduated to oboe, English horn and harp overdubs.

Up until now Miles had only played with the group during the first two days of the session, and then mostly to generate a specific mood. Monday was destined to be his day! He began to jam over the top of the now 1½ hours of recorded material. This was jazz in its purest form, for the most part being performed as "first takes," with Miles seldom stopping to play a particular section again. Later that same day there was a duo performance recorded live, with Davis and the world's premiere acoustic bass player, Niels Henning Orsted Pedersen.

By Tuesday Miles was on his way back to the States and, on the whole, everything went extremely well. Basically, all instruments were routed to the SSL console, except the horns, which first went to the API acting as a submixer. There also was a small sub-mixer out in the studio for the electronic keyboards. (This setup worked well in general, except on one or two electronic occasions when a screaming synthesizer effect came out of nowhere!)

The monitoring system worked very well. Since there were never more than 34 tracks being recorded simultaneously, they could be monitored on the SSL in the usual way. The musicians received the same monitor mix via the foldback cue system, and there were no complaints — the ultimate compliment on any session!

A few repairs were done on Tuesday, but most of the day was spent doing rough mixes and cataloging information.

With the composer out in the studio, and no one else in the control room being familiar with the piece, there was a problem keeping track of all the different movements, titles, tracks, etc. The SSL Studio Computer proved to be a great help in storing all this information after the session. Palle took home rough mixes of everything, and booked time a week later for more repairs and overdubs. At this time he also did some overdub playing of his own style of trumpet.

Mixing was pretty straight ahead, with both Niels-Erik and Henrik Lund continuing the team engineering concept. Palle also got in on the act, with the SSL computer acting as the fourth member of the "team." The final stereo mix went to a 30 ips analog machine, with a Sony PCM-1610 running in parallel.

Now, things are back to normal at Easy Sound (whatever that means), and the 1½ hours of music have been edited to a double album. But for Palle, Niels-Erik, Henrik, the musicians, and everyone else in the home of Hans Christian Andersen, the days Miles came to town to "blow" will not soon be forgotten. ∎∎∎

ENSONIQ MIRAGE DSK-8 DIGITAL SAMPLING KEYBOARD

Reviewed by Terry Fryer
Colnot-Fryer Music, Chicago

Since the introduction of the Fairlight Computer Music Instrument in 1979, sampling keyboards have remained pretty much in the rich man's realm. While E-mu Systems slashed the price drastically of such systems with the introduction of the Emulator I, basically these instruments still have remained out of reach of the majority of musicians and studio operators. In 1982, the team responsible for the design development and manufacturing of the Commodore 64 personal computer decided to transfer their philosophy to the music industry; thus was born the company of Ensoniq. One of the first products to come from Ensoniq was the Mirage Digital Sampling Keyboard.

The Mirage is an eight-voice sampler that is complemented by a complete analog synthesizer section. It features a 61-note keyboard with left-hand pitch and modulation wheels. A control grouping on the front panel consists of a volume slider; a control section of four buttons with accompanying alphanumeric display; a telephone-like keypad for selecting functions; four multi-purpose buttons for sequencer and MASOS (Mirage Advanced Sampling Operating System) control; and two groups of two buttons for floppy diskette access and sampling. The rear panel contains audio-in (switchable between mike-and line-level), audio-out, twin MIDI jacks, external sync input, footswitch jack, expansion port, and the power on/off switch.

A sampling keyboard takes an external signal, whether from a micro-phone- or line-level input, and converts it into a form that the internal microprocessor can understand. The digitized sound can then be manipulated internally and converted into a signal for connection to an external playback system. With the basic Mirage, the steps between plugging in a sound source and playing it back as a sample are short and easy. Ensoniq has also taken the liberty of offering several accessories with the Mirage for the serious sampler, which include an Advanced Sampler's Guide, an extended sampling filter, and a complete Visual Editing System (VES).

Ease of Operation

On an operational level, the Mirage can be up and running in about 15 seconds using microfloppy diskettes from the unit's sound library. A total of sounds with four programs each, plus eight 33-note sequences, can be stored on each 3½-inch microfloppy. Samples stored on a diskette are accessed by pressing the Load buttons, and the desired number — 1, 2 or 3 — on the control panel. The keyboard is divided into an upper and lower section, with controls that allow sounds to be loaded from diskette into either the upper half, the lower section, or the entire keyboard. Given the small size and convenience of the 3½-inch diskettes, sounds and their programs can be easily transported from machine to machine.

Most users will find that the real fun of the unit comes from modifying the aforementioned sounds. As noted above, the Mirage features a complete analog synthesizer section, which can also provide some interesting moments in itself. Functions for the analog section are accessed by a two-digit code entered into the "Select" keypad. An LED display shows the present function being worked on, or its value. The values are changed by using the Up and Down arrow keys in the control section; the manufacturer has thoughtfully included a small plastic card upon which are printed the commonly used parameters listed by name and number.

When a sound is loaded from diskette, the wave samples are placed in two digital oscillators. If the same sound is loaded into both oscillators (parameter #28-off), then the second oscillator can be de-tuned (parameter #33) to create extremely fat flanging and chorusing effects. Conversely, if different waves samples are loaded in via parameter #28-on, the modulation wheel, velocity sensitivity (parameter #35) and an internal mixer can then be utilized to mix or fade between the two sounds. The guitar library diskette makes good use of this feature, with the modulation wheel controlling the amount of feedback; played loud or through an amp, the resultant sound is amazingly authentic.

The voltage-controlled filter is a four-pole, lowpass variety used to add interest to sampled sounds. By adjusting the cutoff frequency (parameter #36), resonance (#37), keyboard tracking (#38), envelope (attack: #40; peak: #47; decay: #42; sustain: #43; release: #44), applying a little keyboard scale decay (#47) and velocity sensitivity to the filter peak (#46), the lower part of an acoustic piano can be turned into a truly *monstrous* synthesized bass.

Similar functions are found in the unit's digitally-controlled amplifier (DCA). Take the same acoustic piano; a little care with the envelope (attack: #50; peak: #51; decay: #52; sustain: #53; release: #54), some keyboard velocity sent to the attack portion (#55), and release time (#59), an "ethereal," string-like texture can be achieved.

For an "Eddie-Van-Halen-meets-Van-Cliburn" effect, I also found it irresistible to apply a little LFO (frequency: #31; depth #32) and pitch bend (range: #22) to create a spring-loaded, tremolo Steinway sound.

Monophonic mode (#29), master tune (#21), complete modulation of the filter and amplitude envelopes by velocity and keyboard scaling, footswitch control for the sequencer and sustain (#89), keyboard velocity sensitivity (#23) and keyboard balance (#24) round out the Mirage's performance features.

Perhaps the only thing that takes a little explanation is the top-key parameter (#72). It is possible to have one to eight wave samples present in each

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half of the keyboard at any one given time. There is a priority scheme for how these wave samples are assigned: the lower half takes priority over the upper, and in each half they are ranked from low to high. Parameter #72 defines the top key for each of the voices, and can be used to construct eleborate splits of up to 16 (eight per half) wave samples. The rest of the parameters control the MIDI assign-

SUMMARY OF ENSONIQ MIRAGE DIGITAL SAMPLING KEYBOARD SPECIFICATIONS

Keyboard: 61 keys (C1 to C6) digitally encoded with velocity sensing, eight voice multisplit keyboard for polytimbral performance and sequencing.

Dimensions: 43.25 by 12 by 3.25 inches (W×D×H).

Disk Drive and Disks: low-profile disk drive with high-density, 3.5-inch microfloppy format, 400-Kbyte storage capacity provides each disk with three full-keyboard sounds composed of up to 16 wave samples per sound (up to 48 wave samples per disk), four programs for each upper and lower memory segment, per sound (up to 24 programs per disk), eight sequences of up to 333 events each (three sequences of up to 1,357 event with Sequencer Expander Cartridge), maximum load time of four seconds per keyboard half.

Sound Library Diskettes: #1 — Piano, bass, fuzz guitar, synth bass, percussion, wood flutes; #2 — Synthesizers; #3 — Cellos, violins, upright bass, saxophones, trumpets, trombones; #4 — Rock drums, electronic drums, orchestral drums. (includes snares, basses, toms, cymbals, tympanies, timbales, handclaps, gongs, etc.); #5 — Piano, (plus one octave), marimba, electric piano; #6 — guitars. (Note: #1 and #2 are shipped with the Mirage.)

Sampling: 16-bit (96 dB) dynamic range; output response to 14.7 kHz; variable sample rate from 33.3 to 8 kHz (sample rate expandable to 50 kHz with optional Input Sampling Filter); variable sample times from two to eight seconds per memory segment; single-sampling (wave sample per memory segment); multisampling (up to eight different wave samples per memory segment); wave samples can be assigned to programmable keyboard zones; adjustable looping with resolution of one byte; adjustable pitch, volume and timbre of individual wave samples; variable four-pole input anti-aliasing filter (optional seven-pole variable Input Sampling Filter); selectable mike- or line-level input.

Programs: up to four programs per upper or lower memory segment; programs can be called up instantly (no loading time) and control envelopes, filters, modulation and effects; programs can be duplicated, linked, or transferred between keyboard halves.

Oscillators: 16 digital oscillators provided by custom Q-Chip™; no tuning on power-up or periodic retuning required; programmable master tune variable over a five semitone range in 0.05-semitone steps.

Filters: eight independent voltage-controlled lowpass filters; cutoff frequency range 50 Hz to 15 kHz; programmable resonance (Q); 24 dB per octave (four-pole) rolloff slope; programmble keyboard tracking; automatic filter tuning.

Envelopes: 16 programmable envelopes per voice (filter modulation envelope for synthesizer effects and amplitude modulation envelope for altering dynamics); each envelope has five programmable parameters (attack, peak, decay, sustain and release) each of which has independently programmable velocity modulation; keyboard scaling of decay rate.

LFOs: eight independent, retriggerable LFOs; modulation depth can be stored in a program or varied by the modulation wheel; LFO frequency variable from 0.5 to 40 Hz; modulation depth from 0.05 to 12 semitones.

Effects: two oscillators per voice for chorusing or flanging; layering wave samples provides two sounds per voice; balance between sounds is controlled by either key velocity or modulation wheel.

MIDI: current published IMA V1.0; selectable Omni or Poly mode (1 to 16 channels); pitch bend and modulation wheels transmitted and received; MIDI Clock in/out allows syncing with external MIDI drums and sequencers; selectable MIDI out or Thru modes.

Polyphonic/Polytimbral Sequencer: 333 events/notes (expandable to 1,357 with optional Sequencer Expander Cartridge); records keys, key velocity, pitch bend, modulation and sustain pedal; records external MIDI data; overdubbing within eight-voice capacity; playback tempo variable from half to four times speed.

External Sync: Allows syncing to non-MIDI drum machines or sequencers; automatically quantizes Mirage sequencer to any external clock; external clocks are also sent out to MIDI, allowing other MIDI devices to sync to a non-MIDI source.

Wheels: pitch bend and modulation/mixing.

Suggested List Price: Model DSK-8 $1,695, with two sound disks; FD-1 blank formatted microfloppies $12.95; ASG-1 Advanced Sampler's Guide with MASOS disks $49.95; SQX-1 Sequencer Expander (1,024 events) $69.95; ASP-1 Apple IIe Visual Editing System $299.95; ISF-1 Input Sampling Filter (enables sample rates up to 50 kHz) $149.95; Digital Multi-Sampler $1,395; system now includes a disk-formatting diskette free of charge.

Manufacturer: Ensoniq, 263 Great Valley Parkway, Malvern, PA 19355. (215) 647-3930.
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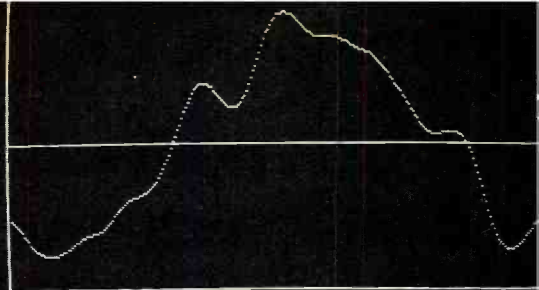
WAVEFORM DISPLAY: PAGE ▓▓ LOCATION ▓▓
ARROWS:LEFT/RIGHT-SCROLL UP/DOWN:PAGE
<V> VIEW COMMANDS * <ESCAPE> TO EXIT

Sample Waveform Display

ments, and sampling and diskette commands.

Controls on the Mirage's sequencer are fairly straight forward: a four-button section on the control panel, combined with five parameter functions from the select keypad, handle all the chores. Eight, 333-note sequences can be stored and loaded from disk, along with the sound files and program parameters. An expansion cartridge is available to increase sequencing capacity by 1,024 events. For convenience, the sequencer also accepts external clocks via MIDI or the sync jack (#86), has the ability to loop a sequence (#88), vary the playback speed (#87), and start and stop from a footswitch (#89).

External Sampling Capability

Ensoniq's Advanced Sampler's consists of a bound manual and two disks that contain the Mirage Advanced Sampling Operating System (MASOS). The manual provides a more thorough description of the Mirage than that in the Mirage Musician's Manual, and also contains a good introduction to the art of sampling, plus some helpful tips on sampling with the Mirage.

When MASOS disks are booted, the sequencer's Record and Play buttons take on the functions of selecting upper and lower wave samples, thus facilitating rapid switching between various samples. The Load sequence button becomes the MASOS special-function key and, in combination with parameter functions 86 and 96, provides the following functions: copy data; fade in; fade out; scale with a linear ramp; add; invert; reverse; and replicate — the types of features found previously only on megabuck digital synthesizer systems. Unfortunately, keeping track of all of this information through a small two-character alphanumeric LED display can be quite trying. Fortunately, however, this portion of the system is complemented with the addition of the Mirage Visual Editing System. To utilize MVES, you need an Apple IIe or II+

ENSONIQ MIRAGE DIGITAL MULTI-SAMPLER

The rack-mountable Digital Multi-Sampler can be connected to any MIDI-equipped synthesizer, keyboard or other controller. In addition to offering the sound of digitized acoustic instruments, it also responds to pitch bend, modulation, aftertouch and breath control effects sent over MIDI channels. The unit has a 61 semi-tone range (five octaves), and can be programmed with over 30 playback parameters including filter and amplitude envelopes, modulation and dynamics. It also features an on-board sequencer with overdub capabilities.

Sounds and programs stored on 3½-inch diskettes are loaded into the Mirage from a built-in drive. The Ensoniq Sound Library currently consists of over 100 sounds and programs, featuring keyboard, string, brass, reed and fretted instruments, plus many special effects.

For sampling external sounds, the unit's 128-Kbyte can accommodate up to 16 distinct samples at one time; each sample can be played back polyphonically through the Mirage's eight voices.

According to Rob Weber, Ensoniq marketing director, the new device differs from many "add-on" sampling units in many ways — the most important of which is its ability to multi-sample. "Synthesizers use simple waveforms which transpose up and down the pitch scale with relative ease. Most acoustic instruments, however, are made up of very complex waveforms which can only be raised or lowered in pitch by a few semi-tones before losing their character. Multi-sampling allows instruments to be sampled in small pitch increments, preserving the character of the sound over a number of octaves," he offers. □□□

OPERATIONAL ASSESSMENTS — ENSONIQ DSK-8 REVIEW —

with 80-column card, but it's still a great bargain no matter how you look at it.

MVES makes all of the normal Mirage controls, plus the MASOS data manipulation commands available on the Apple IIe or II+. Since data MIDI Exclusive commands, some attention must be paid to MIDI cabling between the computer and the Mirage. In total, seven pages accessible from the main menu break the commands into usable groups:

•**Waveform Display** could best be defined as a graphic representation of the wave samples in the Mirage memory (Figure 1). Each half (upper and lower) have 64 Kbytes memory divided in 256 "pages," a page being 256 points across and 128 high. (There are actually 256 values, but they won't all fit on the Apple IIe screen.) The commands are easy to use, and include several modes of display for the waveform, as well as a number of means of moving throughout the waveform and to other parts of the MVES system. With the display, it is very easy to judge whether or not you have sampled at the correct amplitude, and it is also quite obvious when you have gone into clipping, or are experiencing a normal situation. The Draw Waveform command utilizes a cross hair controlled by an external graphics controller, or the cursor keys of the Apple, to re-draw the waveform.

•**Wavesample Parameter Display** provides a representation of the parameters related to individual wave sample control (parameters 60 thru 72), as well as the positions of the loop pointers, relative tuning, relative amplitude and filter frequency, and the value of the top key. It's easy to see at a glance the status of a particular wave sample, adjust it, and move on to the next sample.

•**Program Parameter Display** shows the parameters (#28 thru #59) that are associated with the programs. All of the filter, amplifier, envelope and envelope-modulation parameters are visible at one time, and allow a quick grasp of why a program sounds the way it does. Again, any of the parameters can be altered from either the Mirage, or the Apple Keyboard.

•By utilizing the **Configuration Parameter Display**, you can also view the keyboard, MIDI and sampling configuration parameters with a quick glance.

•Since there can be up to eight wave samples stored at one time in each half of the Mirage memory, **Wavesample Memory Map Display** sorts

Apple Visual Editing System

out which wave sample is where. Looping addresses are also displayed, as well as the top keys of each wave sample, making complex keyboard splits a snap.

•**MASOS Data Manipulation Functions** make available all of the functions described in the MASOS section, with the addition of a modified rotate command. Easily accessed from the waveform Display page, this page is additionally equipped with a cursor that can be used to set the start and end parameters of the data-manipulation function.

•**Data Transfer From Mirage** is the last of the seven pages contained within the visual Editor. Basically, it loads data from the Mirage, and keeps the MVES from becoming "stupid" and not reflecting the present condition of the Mirage memory, in the event that the MIDI cables come unplugged, or the external computer enable switch is turned off.

Overall, because of the design of the Mirage's custom-designed Q-Chip™ (Bob Yannes, Albert Charpentier, and Bruce Crockett developed the Ensoniq Mirage around a VLSI chip of their own design, which they named the Q-Chip), sampling on the instrument can turn into quite an adventure —because what goes in is not *always* what comes out. However,

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LATEST DEVELOPMENTS:
Mark II Version of DSK-8 Mirage, Plus new Operating System and Sound Lab Editing Software for Apple Macintosh PC

According to Robin Weber, Ensoniq's director of marketing, the company has now released several enhancements for the DSK-8 and Multi-Sampler:

• The latest version of the DSK-8 includes a new keyboard, circuit changes to enhance noise performance, and one or two cosmetic upgrades.

• A new enhanced operating system that accommodates additional MIDI commands, including pressure and breath-controller information from, for example, a Yamaha DX-7, and the ability to load new voices from disk via an external keyboard.

• Sound Lab Editing Software for the Apple Macintosh, developed by Blank Software, which enables sampled sounds to be edited and manipulated using the "user-friendly" features of a Mac PC, including icons and pull-down menus, plus screen dumps to a suitable dot-matrix printer.

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for the price, it's hard to complain and, besides, the Advanced Sampler's Guide and Visual Editing System Manual are full of tips that can help place your favorite rude noise into the units.

There are also a few final points worth mentioning about the Mirage: this unit is a "Rock 'n Roll" sampling keyboard: plug in any signal, push a few buttons, and you're in business. Also, the power of the analog synthesizer section on a sampling keyboard should never be underestimated, since sounds in this domain can be manipulated with such speed — when compared to digital methods — that it becomes frightening.

As for drawbacks, the Mirage ocasionally produces some unwanted noise, which means care must be exercised when considering methods of recording; newer units contain a compansion circuit that lessens this problem, and a simple modification is available for older versions. In my experiences with the Mirage, I found that the use of noise gates, judicious equalization, and external signal processors such as the Aphex Aural Exciter improved the unit's output significantly.

The newest addition to the line is the rackmountable Mirage Digital Multisampler — see accompanying sidebar for further details — which comprises a complete Mirage *sans* keyboard. Just the thing for that "hard-to-buy-for" friend. ∎∎∎

SENNHEISER MKH40-P48 STUDIO CONDENSER MICROPHONE

Reviewed by Lowell Cross, Professor of Music and Director of Recording Studios, University of Iowa

For over 20 years, Sennheiser has been identified with solid-state condenser microphones, based on the radio-frequency oscillator/detector (RF modulator/demodulator) principle. Since this engineering concept for electrostatic transducers antedates field effect transistor (FET) designs by several years, Sennheiser can lay claim to being one of the *very* first manufacturers of solid-state microphones. Well known examples of the RF design have been the MKH805, MKH815, and MKH816 "shotgun" microphones, which are in widespread use for radio and TV news events, sound reinforcement, and general applications for the motion-picture industry.

RF condenser models are highly respected for their very low self-noise output. Briefly stated, the condenser capsule is the variable element in a crystal-controlled oscillator operating at 10 MHz. High-Q circuits of this type yield extremely low noise; after demodulation, the audio output from the circuit may be amplified by conventional means (see pp. 183-186 of Howard W. Tremaine's *Audio Cyclopedia*, second edition, for a more complete description).

In spite of their admirably quiet operation, other factors have tended to tarnish the reputation of RF microphones, in comparison to either tube or FET types. My experience with the Sennheiser line extends back to 1972, when MKH105s (omni) and -405s (cardioid) were installed by the sound-reinforcement contractor in our 2,600-seat Hancher Auditorium. (Since then, they have been replaced by another brand.) I do not think that I am alone in observing the characteristically bright, even harsh, qualities of these models from the recent past, especially when they are used to record most forms of music. Furthermore, the earlier MKH units required A-B or "modulation-lead" powering (DIN 45 595), which is incompatible with the 48-volt "phantom" system in common use today (DIN 45 596). I am pleased to report that with the introduction of the MKH40-P48, Sennheiser has overcome virtually all of the limitations of their previous RF microphones.

The MKH40-P48 differs from its predecessors by the use of 48V powering, a symmetrical capsule (fixed cardioid), and additional operating features: zero or –10 dB attenuation; switchable 6 dB per octave rolloff characteristic below 120 Hz). Furthermore, the output circuit is transformerless. Our review models were finished in matte black; in general, these

small units are very attractively styled and crafted. External dimensions are 25mm (one inch) in diameter by 150mm (six inches) long. The price announced by Sennheiser is $685, placing it in a competitive range with the AKG C414EB/P48 or the Schoeps CMC54U.

Evaluation Sessions

To evaluate the new Sennheiser models, we used them to record a concert given by the University Symphony Orchestra on September 25, 1985, in Hancher Auditorium. The program consisted of: Haydn's *Symphony No. 104* in D Major ("London"); Debussy's *Iberia*; and Beethoven's *Concerto No. 5* in E-Flat Major, op. 73 ("Emperor"), with James Dixon conducting, and Kenneth Amada, piano soloist. The two MKH40-P48 microphones were mounted as a near-coincident pair with their capsules about 17 to 18cm (approximately seven inches) apart, each angled 45 degrees away from the center line of the orchestra.

For purposes of comparison, a pair of Neumann TLM170s were mounted slightly under the Sennheisers (3 to 4cm, or one to two inches below) at exactly the same angle and distance apart as the Sennheisers. The TLM170s, which have been our control microphones throughout all of our evaluation sessions for *R-e/p*, were set in the cardioid position. No low-frequency rolloff or capsule attenuation was employed for either type.

The four microphones were suspended 3.6 meters (12 feet) above the stage, and 3 meters (10 feet) downstage from the conductor's podium. This arrangement placed them slightly below the line defined by the fully-raised lid of the Steinway nine-foot grand piano, which was moved onstage for the concerto after intermission.

Studio equipment and techniques were the same as those used for the evaluations reported in previous issues of *R-e/p*: Neve 5315 24-channel console, ANT Telcom noise reduction systems, Studer A80/VU MkIII 24-track recorder operating at 15 ips with Ampex 456 Grand Master tape, and Klein+Hummel O92 reference monitor loudspeaker systems. Obviously, only four channels were required for this experiment — two each for the two types of microphones. A cable run of 150 meters (500 feet) interconnects the auditorium with our control rooms. Neither microphone pair exhibited any noticeable high-frequency losses when operated over this distance.

The extremely low self-noise characteristic of the MKH40-P48s became evident as we monitored the "ambience" of the quiet, empty audito-

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rium during the afternoon following our setup for the dress rehearsal on September 24. The hall's background noise level, of approximately NC 15 to NC 20, completely masked any self-noise from either set of microphones. But the MKH40-P48s outperformed the TLM170s in one significant respect: they have at least 10 dB more output for a given sound-pressure level. The requirement of less pre-amplifier gain (35 dB versus 45 dB, in this instance) offers the very real

SUMMARY OF SENNHEISER MKH40-P48 STUDIO CONDENSER MICROPHONE SPECIFICATIONS

Directional Pattern: Cardioid.

Acoustic Operating Principle: Symmetrical pressure gradient.

Frequency Response: 40 Hz to 20 KHz.

Sensitivity: 25 mV/Pa (8mV/Pa), ±1 dB.

Nominal Source Impedance: 150 ohms, balanced.

Minimum Load Impedance: 1 kohms (600 ohms to 130 dB).

Equivalent RMS SPL (DIN45 500): 12 dBa (16 dBa).

Equivalent Peak SPL (CCIR 468): 12 dBa (26 dBa).

Maximum SPL for less than 0.5% THD at 1 kHz[1]: 134 dB (142 dB).

Rolloff Frequency: *Low-cut out* is −3 dB at 40 Hz, 12 dB per octave; *Low-cut in* is −3 dB at 120 Hz, 6 dB per octave.

Weight: 100 grams.

Dimensions: Diameter is 25mm; length is 150mm (one inch by sic inches).

Price: $685, suggested list.

Note: 1. Measured with transducer capsule on microphone.
Bracketed values indicate pre-attenuation "in."

Manufacturer: Sennheiser Electronic Corporation
48 West 38th Street, New York, NY 10018. (212) 944-9440

POLAR RESPONSE

125 Hz		2000 Hz	——
250 Hz		4000 Hz	—·—
500 Hz		8000 Hz	—··—
1000 Hz		16000 Hz	—···—
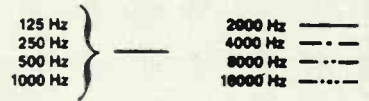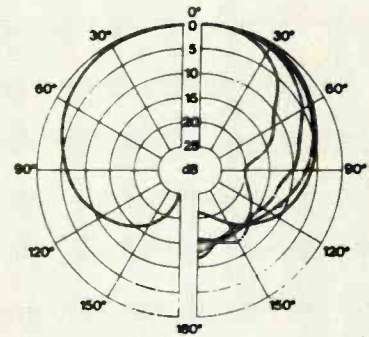
SENNHEISER electronic
* 10079
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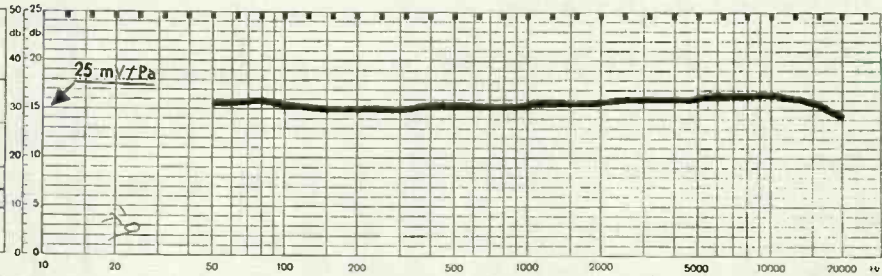
SENNHEISER electronic
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Typ: MKH 40 P 48 U - 3
Prüffeld Pot: 25 dB / 50 dB
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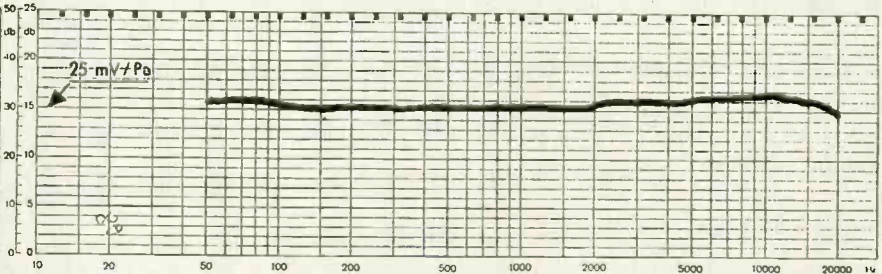
Copy of actual documentation that accompanied evaluation microphones.

benefit of less noise from the input electronics — no matter how quiet the mixer or console might be. Since the peak levels at the microphones during the concert were in the SPL range of 100 dB, we were unable to compare the relative performance of the two types in extremely loud environments. For this evaluation, neither the Sennheisers nor the Neumanns were being operated anywhere near their respective overload regions.

The monitoring assignments in the console permitted direct A-B comparisons between the two pairs during the concert without disturbing the recording. Our audio engineer, Peter Nothnagle, and I, therefore were able to compare the audible qualities of the two types without listening through the tape medium. During a subsequent playback session, we were joined by another engineer on our staff, David Muller. The three of us formed the "audition panel" for this report; unfortunately, there was not enough time to assemble a larger group of listeners. I personally have evaluated these microphones by listening to the entire concert twice, once "live" and once during playback of the multichannel tape.

Subjective Appraisal

The three of us completely agreed that these particular MKH40-P48s were perceptibly "brighter" in quality than the TLM170s. In all other respects, the Sennheisers are hard to fault. The capsule design indeed appears to be capable of very low distortion and a good cardioid pattern across the audible range. Other than the "bright" quality already mentioned, no unusual colorations were evident, either in the direct sound from the orchestra, or in the rendering of the hall reverberation. It should be noted that the Neumann literature for the TLM170 indicates a slightly receding upper-midrange response between 2 and 7 kHz when the microphone is set for the cardioid pattern (-2 dB at 4 kHz). With the response of the MKH40-P48 rising in this same frequency band, the "colorations" of the two types were clearly different — and audible — in our A-B listening tests. As I have acknowledged many times before, such evaluations require subjective, personal judgements on the part of the listener.

Many engineers prefer a "bright" sound as a form of coloration to enhance their recordings, so a rising high-frequency response from a microphone is not *always* viewed as a limitation. However, some mention must be made of the discrepency between Sennheiser's advertised claim for "ruler-flat frequency response" and the outcome of our listening evaluations which, in fact, were confirmed by the calibration curves supplied with the microphones on loan. (See accompanying line drawing of calibration curves for MKH40-P48 microphones, serial numbers 10079 and 10080.) The response curves for these two units exceeded the published toleranced by rising 2 to 3 dB in the 6 to 10 kHz range. Since we heard a perceptibly "thin" quality in the orchestra's string tone, and a somewhat "tinkly" effect in the piano sound from the Sennheisers, in comparison to the Neumanns, we concluded that the two calibration curves published here are reliable.

In conclusion, the new Sennheisers produced a clean "condenser sound," far preferable to that of earlier MKH models. I, for one, would be most interested in using a pair of truly "ruler-flat" MKH40-P48s to record a symphony orchestra — or any other musical ensemble — in stereo. With the prospect their availability, we can look forward to a new and highly desirable series of RF microphones bearing the Sennheiser trademark.

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Author's note: I wish to thank Anthony D. Cafiero, product manager for Sennheiser Electronic Corporation in New York, for loaning us the microphones used in this evaluation — *LC.*

www.americanradiohistory.com

ENSONIQ INTRODUCES ESQ-1 DIGITAL SEQUENCER/SYNTHESIZER

The ESQ-1 is an eight-voice polyphonic, polytimbral synth with three oscillators per voice. A choice of 32 multisampled and synthetic waveforms is available, including sampled waveforms of piano, strings and brass instruments, in addition to a wide variety of synthetic waveforms.

An 80-character fluorescent display shows 10 programs by name. Fourty on-board programs with an additional 80 cartridge programs are available.

The ESQ-1 features a polyphonic, velocity-sensitive 61-note weighted-action keyboard with programmable split points and sound layering on either or both keyboard halves. MIDI features include a special MIDI Overflow Mode that permits slaving other MIDI units together to create a 16-voice synthesizer. There are also Poly, Omni, Multi and Mono modes, plus eight simultaneous polyphonic channels with separate programs.

The ESQ-1 contains a sequencer with 2,400-note internal storage (expandable to 10,000 notes) and eight discrete tracks, each with separate program and MIDI channels; each track has eight voices dynamically assigned.

ENSONIQ CORPORATION

For additional information circle #194

NEW MIDIVERB DIGITAL REVERB UNIT FROM ALESIS

Supporting 63 different pre-programmed setups, the stereo in/out Midiverb provides short (0.2-second) and long (20-second)
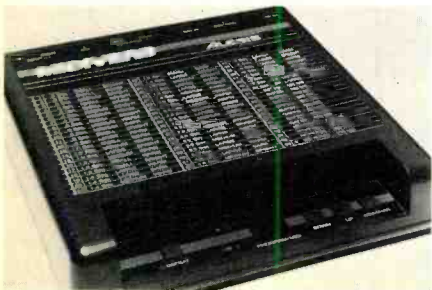
decay times with a variety of sizes and tonal textures. For special effects, the Midiverb offers nine gated and four reverse reverb programs. Full MIDI control is also featured.

The new unit uses a new design approach, RISC (Reduced Instruction Set Computer) Architecture, which is said to provide the programmer with a very small set of possible instructions that execute very quickly, as opposed to more complex machines that offer more apparent power but operate slowly. The Midiverb uses an extremely small instruction set that runs at three million instructions per second, outperforming most systems on the market today, the company claims.

Specifications include a frequency response of 20 Hz to 10 kHz, and dynamic range of 80 dB, typical.

Manufacturer's suggested retail price of the Midiverb is $399.

ALESIS

For additional information circle #195

TWO NEW MIDI PRODUCTS FROM J.L. COOPER

• MidiLink is a rack-mount "MIDI program manager" that allows a master keyboard to send different MIDI program numbers to individual slave synths simultaneously. The unit will store up to 99 master programs that can then be recalled from any MIDI keyboard or sequencer. A master program may contain up to 12 separate MIDI program change commands, each of which may be sent out one of six rear-panel connectors and on a desired MIDI channel number. In addition, up to 32 master programs can then be linked in a chain, and stepped forward or backward by means of a footswitch; up to 100 chains can be stored in battery-backed memory. Suggested retail price is $449.

• Midi Mute, the first product in the company's MidiMation Series, is an eight-channel MIDI-controlled muting device for audio mixers. Designed to be connected between instruments (or effects) and mixer inputs, the unit generates and receives MIDI Note-on and -off commands, and converts them to switch closures for its internal relays. In addition, any channel may be used as an isolated switch closure to simulate a footswitch. The device comes equipped with a hand-held remote controller that connects to the main rack-mount unit through an ordinary modular telephone cable, and is expandable up to 24 channels by attaching expander units; all

24 channels can then be operated from the remote controller. Suggested retail price is $549, including the remote controller; eight-channel expanders will carry an SRP of $295.

J.L. COOPER ELECTRONICS

For additional information circle #196

DIGIDESIGN SOUND DESIGNER VERSION 1.1 FOR EMULATOR II

Version 1.1 of Sound Designer for the EII and Macintosh includes many new features and performance improvements. The program's speed and ease of use are said to have been dramatically improved: digital processing functions such as digital mixing, gain changing and merging of waveforms are now described as running up to 10 times faster than version 1.0. New features include digital equalization, crossfade looping and enhanced digital synthesis.

• The Digital EQ program provides fully parametric peaking EQ, hi/lowpass EQ and hi/low shelving EQ. After the EQ settings are entered, a frequency response graph of the digital filter can be displayed on the Macintosh screen.

• The Crossfade Looping feature is said to be very useful for creating click-free loops in any sound. It uses a complex pattern of crossfades to mix sound data around the loop start with data around the loop end.
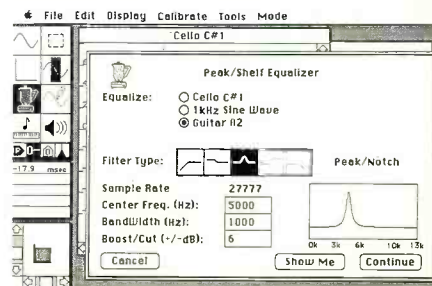
• Many parameters have been added to the Karplus-Strong Digital Synthesis program. Harmonic type and content, pluck position, resonance and other values can be accurately controlled. The synthesis program can even be fed by a sampled sound, creating some very interesting and unusual effects similar to vocoding and reverb, the company says.

In addition to these new programs, many new waveform editing tools have been added. Loops can be played using the Mac's internal sound driver and speaker, and any portion of a sound can be selected and played separately on the Mac or the Emulator II.

Version 1.1 has been shipped free of charge to all registered Sound Design owners.

DIGIDESIGN, INC

For additional information circle #197

360 SYSTEMS ANNOUNCES MIDI ROUTING SYSTEM

Midi Patcher is a four-input, eight-output MIDI routing system that allows the user to program up to eight routing configuration and store them in battery-backed memory.

Configurations can be recalled via the front-panel buttons, or via MIDI program change command.

Eight groups of four different colored LED's on the front panel allow the user to see the current MIDI routing at a glance. A test button sends a short MIDI sequence over the selected channel to verify MIDI continuity.

Packaged to occupy one space of a standard 19-inch rack, suggested retail price of the Midi Patcher is $295.

360 SYSTEMS

AUDIO PRECISION ADDS ENHANCED INTERMODULATION TESTING TO SYSTEM ONE MEASUREMENT UNIT

DIM and TIM (dynamic intermodulation and transient intermodulation) testing by the mixed squarewave-sinewave method has been added to the established mixed sine-wave methods of SMPTE, DIN, and CCIF difference tone. According to Audio Precision, DIM and TIM tests have attracted increasing attention in recent years as engineers explore system performance near slew-rate limits. Some experts suggest DIM/TIM phenomena as the reason why two audio systems sound differently, even though they have virtually the same measured performance according to older, more static test methods.

System One's implementation of DIM/TIM testing is said to allow measurement of distortion at levels typically 20 dB lower than spectrum analyzer measurement capability.

IMD tests can be selected and run even by semi-skilled staff, as can other measurement parameters of System One, including total harmonic distortion, quantization distortion or noise, phase, broadband or third-octave selective level and noise, and frequency. Since the unit is PC-based and graphics oriented, sweeps versus frequency or amplitude are made in seconds, and graphically plotted on the computer screen as they are made. Results may be saved to disk or graphically printed by an inexpensive dot matrix printer.

The new complete intermodulation distortion option is priced at $1,200. A complete System One with dual-channel generator, dual inputs, and the complete measurement capability outlined above is priced at $7,250, including software.

AUDIO PRECISION

AUDIO INTERVISUAL DESIGN INTRODUCES ALBRECHT MB-51 MAG RECORDER TO U.S. MARKET

The sprocketless MB-51 utilizes capstan drive and microprocessor control. Currently in use at over 700 film and television facilities throughout Europe, the MB-51 is said to meet all requirements of modern studio practice, and can be used flexibly in post-production, dubbing and mixdown studios with projectors, editing table, film scanners and VCRs.

A plug-in head assembly accommodates 16mm, 17.5 and 35mm mag film, and the system handles all standard formats from mono through six-track, with multiple equalization pre-sets included. Film speed can be continuously controlled up to a maximum of 750 fps

for 16mm, and 300 fps for 17.5/35mm mag film. Since the system is entirely servo-controlled, wow and flutter specs are said to be exceptionally low. An electronic film counter reads out in feet, meters or in minutes, seconds and frames.

AUDIO INTERVISUAL DESIGN

For additional information circle #201

ROLAND MC-500 MICRO COMPUTER MIDI SEQUENCER

The MC-500 has a 40,000-note storage and features a 3.5-inch disk drive. Performance data can be loaded and overdubbed from MIDI instruments in real- step-time, and then stored on disk.
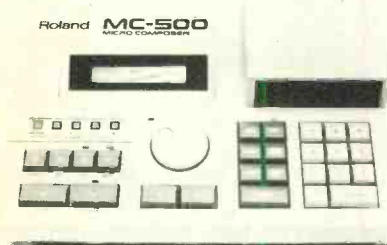
Four tracks are provided for note entry, plus a fifth rhythm track for MIDI drum programming that stores 999 bars and 95 rhythm patterns. The tempo of an entire song can be programmed using the tempo track. Multi-track recording is also possible by merging two or more tracks together, and the MIDI channel assignment of all original track data is retained after tracks are merged. In addition, individual tracks can be edited after they have been merged together.

Punch-in and -out is possible even in the middle of a bar, and individual notes or entire bars of music can be inserted and deleted. Gate time, duration, note number, and velocity can also be altered, plus copying, inserting, deleting, erasing, transposing, and MIDI channel shifting functions from any bar for any number of bars. Songs can be stored in eight individual song areas in the unit's memory, and then chained together.

MIDI information (including System Exclusive) can be inserted at any point in the music and edited from the unit's keypad. The MC-500 can send and receive MIDI Song Pointer Data, which will allow "chasing" to video or multitack tape with SMPTE timecode when used in conjunction with the SBX-80 Sync Box. The unit also features two MIDI outputs that can be assigned to any or all MIDI channels.

ROLAND CORP US

For additional information circle #203

SYNTECH UNVEILS MIDI CONVERTER FOR RHODES CHROMA KEYBOARD

The new Chroma to MIDI Converter is a complete MIDI implimentation residing in a small black box that plugs into the back of the Chroma. No external power supply is required.

Features include 16 programmable MIDI functions set from the Chroma front-panel and saved in battery backup memory inside the Chroma; MIDI System Exclusive for loading and saving patches; reception on up to eight MIDI channels simultaneously (taking advantage of the Chroma's multi timbral feature); complete compatibility with the Fender/Apple Music System, making it possible to convert the system into a MIDI sequencer.

The Chroma to MIDI Converter has a suggested list price of $349.95

SYNTECH

For additional information circle #204

RACK-MOUNT STANDS FROM ULTIMATE SUPPORT SYSTEMS

The new units allow rack equipment to be mounted to suit any performing style in the studio, or on stage. Three basic support options are available in two different sizes.
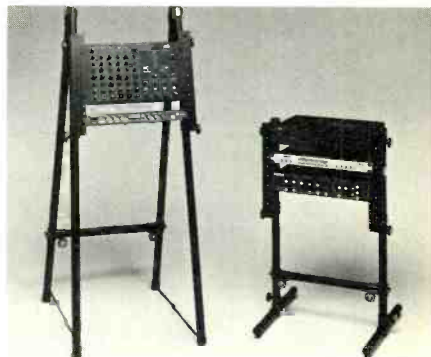
A-frame configurations offer independent support for users with extensive rackmount needs. The T-leg systems is said to provide a practical method for the cost-conscious musician, while Rack Extensions enable performers to integrate keyboards and rack equipment. Both five- and 10-panel rails are available separately to attach keyboard stands.

Like all USS products, the stands are constructed of black or silver aluminum alloy tubing and rails, with glass-reinforced polycarbonate fittings for the maximum stability.

ULTIMATE SUPPORT SYSTEMS, INC.

For additional information circle #205

HOHNER ANNOUNCES RDM-1000 AND RD-500 DIGITAL REVERB/ECHO PROCESSORS

THe RDM-1000 offers complete programmability of all parameters via MIDI, plus storage into eight memories. Not only can the

memories be recalled through any MIDI system — synthesizers, drum machines, MIDI-sequencers, etc. — but they can also be programmed. In addition, the RDM-1000 features a built-in dynamic noise-reduction unit.

The RD-500 offers eight practical combinations of "quick and simple to operate" reverb and delay presets. Reverb delay, echo delay time, amount of repeat and modulation depth

and speed are fully adjustable. The unit also has a built-in dynamic noise-reduction unit.

HOHNER, INC.

HILL AUDIO UNVEILS SOUNDMIX AND RAKMIX MODULAR CONSOLES

Both the 24/4/2/1 Soundmix and 8/4/2/1 Rakmix consoles feature four auxiliary sends, four band EQ, four auxiliary returns, 100mm faders, 12-way LED and VU metering, send and return patch points throughout, balanced and unbalanced output, direct outputs, and a fully regulated rackmount power supply.
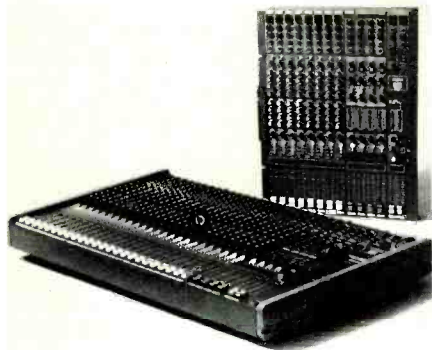
Outputs are selectable +4 dB/ –10 dB, and the auxiliary busses are selectable pre/post.

Suggested retail price of the Soundmix is $3,750, and the Rakmix $2,359.

HILL AUDIO, INC.

DIGIDESIGN SOUND DESIGNER SOFTWARE FOR SEQUENTIAL PROPHET 2000

Sound Designer 2000 allows sampled sounds to be transferred between an Apple Macintosh and Prophet 2000 using a standard Macintosh MIDI interface. Sound data is transferred at twice normal MIDI data rate (63 Kbaud) to minimize waiting, and up to three waveforms are simultaneously displayed on the Mac's high-resolution screen.

The software is said to provide extensive sound editing capabilities, including Mac style "cut-and-paste" editing. The waveform display can be magnified to show fine detail, with editing accuracy to 25 microseconds. Calibration scales provide exact readouts of time and amplitude values at any location in the waveform, and the waveform display can be horizontally and vertically scrolled.

SD 2000 also includes Fast Fourier Transform-based frequency analysis and modification of sounds, digital equalization, digital mixing and digital merging, as well as a variety of other digital signal processing functions for modifying sampled sounds and creating unique sounds. Both sustain and release loops are provided, as well as a crossfade looping program for looping sounds that lack a natural loop area. Direct digital analysis can be performed on the Macintosh, and the resulting sounds transferred to the Prophet 2000 for playback.

Programming of all Prophet 2000 parameters is said to be greatly simplified by the software's graphic programming aids. Filter response curves, ADSR curves, and other graphically represented parameters can be "drawn" using the Mac mouse, and keyboard set-ups, MIDI assignments, controller assignments, etc. quickly programmed using on-screen menus.
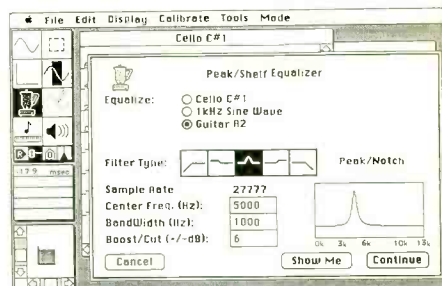
Suggested retail price of Sound Designer 2000 is $495.

DIGIDESIGN, INC.

AMPEX INTRODUCES U-MATIC DIGITAL AUDIO CASSETTE

The new Ampex 467 U-Matic is specially qualified to be free of uncorrected signal errors for exceptional and reliable performance on PCM converters, the company says.

"The Ampex 467 cassette is a product of extensive research into digital machine requirements," explains Bruce Pharr, marketing manager for Audio Tape Products. "Reliable PCM recording requires a cassette with electrical and mechanical characteristics specific to digital audio recording techniques. Ampex uses the correction capability

of PCM converters as a final criteria for the qualification of the 467 Digital Audio Cassette."

All PCM digital audio processors utilize an error-correction system to detect signal errors caused by defects in the recording medium, and electronically correct or conceal them. While small tape defects can be fully corrected electronically and the audio fidelity unaffected, larger defects can be electronically concealed but will adversely affect the audio fidelity. Most serious defects cannot be electronically concealed and cause the signal to completely mute, resulting in an unacceptable recording.

Ampex PCM qualification specifies that there be zero occurrences of error concealment (uncorrectable errors) in each cassette.

The Ampex 467 cassette features a cross-linked copolymer oxide binder system that is said to withstand repeated plays without signal loss or error build up, a feature that allows for extensive editing without exceeding the PCM convertor's digital error correction range. The cassette shell is constructed from high-impact, anti-static plastic to eliminate static build-up, thereby reducing the attraction of airborne contaminants that can cause PCM digital data errors.

AMPEX CORPORATION

For additional information circle #210

SYNTHINET OFFERS SMDS SOFTWARE FOR MUSIC PRODUCERS

Designed to automate music production contracts and job estimates, the two software packages — SMDS Contracts and SMDS Estimates — were developed as a result of a study of musical production houses. "We determined that the automation of contract and job estimate preparation would result in a significant improvement in efficiency and savings in administrative costs," according to Jeff Baker, a studio keyboard technician, and president of Synthinet. "Our software handles the music production paperwork, allowing producers to spend more time being creative."

SMDS Contracts automatically prepares AFM, AFTRA and SAG contracts. Using master files of musicians, singers, contractors, studios, agencies, pay scales, and taxes, the software generates the appropriate contracts and financial reports.

SMDA Estimates calculates musical production costs, including creative, arranging, production, talent, studio, tape, instrument rental, and post production. Various reports, including a job bid letter to the agency, are generated.

SMDS software runs on the IBM-PC and XT, plus compatibles. Synthinet will customize each package to the needs of a particular production house.

SYNTHINET MUSIC DATA SYSTEMS

For additional information circle #211

APL LAUNCHES NEW SERIES OF CONTROL-ROOM MONITOR SPEAKERS

The new series incorporate time-corrected cone and dome driver technologies, and are designed for tonal and stereo imaging accuracy as well as a low-distortion/high-output capability. Models in three-, four- and five-way configurations are available with matching electronic crossover networks. An optional dual 30-inch subwoofer system is available for large control rooms.

As a necessary aspect of achieving accurate monitoring, APL performs on-site set-up and Performance Certification of the monitor systems. FFT and analog B&K instrumentation is used to align driver, crossover network, and amplification parameters.

FFT first-arrival accuracy for the three-and four-way systems is a quoted ±2 dB, 20 Hz to 20 kHz. The five-way system incorporates a ribbon driver to extend the frequency response to 50 kHz. Power handling is 600 watts per channel, with a maximum SPL output of 140 dB at 1 meter.

Prices are from $7,000 to $11,000 per pair, including electronic crossovers, on-site factory set-up, and Performance Certification.

ACOUSTICAL PHYSICS LABORATORIES

For additional information circle #212

MIDI CONTROL SYSTEM FROM IBANEZ

The new system comprises the MIU8 Interface and the IFC60 Intelligent Foot Controller. The MIU8 is a multiple-output MIDI splitter unit, with one MIDI input and eight MIDI outputs, operating on any of 16 MIDI channels.
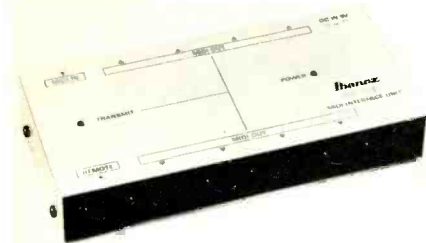
The IFC60 is a remote control with a large three-digit LED readout, and a 15-foot connecting cable. Recall of any of 128 MIDI programs is possible without play interruption, plus control of up to eight MIDI-equipped devices with one main controller.

CHESBRO MUSIC COMPANY

For additional information circle #213

AUDIO LOGIC MT-66 STEREO COMPRESSOR-LIMITER

The new unit provides dynamic range compression from 1:1 to infinity:1, simultaneously accompanied by its own noise gate to ensure quiet operation when no signal is present. Front panel features include a "link" switch to join both compressors for stereo tracking; a five-LED bargraph to indicate gain reduction; gate, threshold, ratio, attack and release controls; plus input and output level controls.

On the rear panel are both balanced and unbalanced inputs and outputs, along with side chain inputs and outputs that can be utilized in changing the compression characteristic, or as a de-esser.

Pro-user price of the MT-66 Stereo Complimiter is $299.95.

AUDIO LOGIC

For additional information circle #214

KURZWEIL UPGRADES K250 SYNTHESIZER AND ADDS NEW OPTIONS

Improvements to the basic instrument include sequencer modifications, new internal sounds, SMPTE capability for film and video scoring, and enhanced tuning of the instrument's 30 resident instrument sounds. In addition, the number of keyboard setups or instrument combinations has been increased from 40 to 97, and a harpsicord sound added to the basic presets.

Modifications to the on-board sequencer memory have increased the basic sequencer capacity from 7,900 to 12,000 notes. In addition, new MIDI synchronization features are said to allow easier interconnection between the K250 and external sequencers and drum machines. The addition of a SMPTE timecode synchronization feature enables the unit to interface with film and video editing

equipment, via a commercially available SMPTE/MIDI converter.

Enhancements can be retrofitted to existing K250s for under $3,000, depending on the model; units shipped from February 1 incorporated the operations software and sound enhancements.

KURZWEIL MUSIC SYSTEMS, INC.

For additional information circle #219

ART RELEASE SOFTWARE UPGRADE FOR DR1 DIGITAL REVERB

The new 1.10 revision software adds 10 new factory presets, bringing the total to 40: Stereo Image, Early Reflections, Gated Snare, Percussive Down Flange, Flanging, and Chorus. Downward flange has been made possible by allowing the Percussive Flange room to have a zero attack time. Other Flanging and Chorus presets use a new room type Flanger/Chorus (FC.1) that allows many different stereo flanging and chorus effects to be produced, with independent control over each channel.

Extended functions include a Demo Mode speed the allows the user to sequence though presets via the K/I switch, and/or a remote footswitch. The number of presets in a sequence, as well as the number and presets, is also programmable.

Planned MIDI features using a manufacturer's ID code to allow the DR1 to inquire and change various parameters, dump and load all of its presets and allow one DR1 to slave from another. In addition to the above, the DR1 may be set such that it can output MIDI program changes when Presets are changed at the DR1.

APPLIED RESEARCH & TECHNOLOGY

For additional information circle #220

TELEX INTRODUCES NEW CD SERIES CASSETTE DUPLICATOR

The new series of high-speed (30 ips) cassette-to-cassette duplicators are available in both mono and stereo versions. Two-cassette "slave" units are available that allow expansion of the single-copy basic unit by up to five slave units for a maximum of 11 copy positions. Slave units include ribbon cable with plugs for easy connection to the master unit or to other slave units. A model with three copy positions is also available.
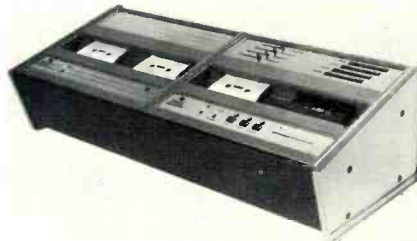
The basic unit includes a function control panel, audio level indicators with slide control, an original cassette position and a copy cassette position. An automatic/manual mode switch allows the user to select one-button automatic operation, or the more selective manual mode. In the auto mode, the user simply presses the copy button and all tapes are rewound to their beginning and copied automatically. When all tapes are finished, they are all rewound again.

TELEX COMMUNICATIONS, INC.

For additional information circle #221

ADAMS-SMITH ACCEPTING ORDERS FOR 2600 A/V DOUBLE SYSTEM EDITOR

The 2600 A/V can be used as a conventional single-system video editor in either a playback/record configuration or an A/B Roll configuration. It can also be used as a single-system audio editor, offering extensive list manangement as well as punch-in and -out accuracy.

In addition, the 2600 A/V can be used as a Double-System, off-line video on-line audio editor. In this configuration, the picture and sound are edited simultaneously, but on separate tapes. Advantages to shooting and editing Double-System are said to include enhanced audio quality, improved flexibility of editing, and lower audio-for-video post-production costs.

ADAMS-SMITH

For additional information circle #222

IBANEZ EPP400 EFFECTS PATCHING UNIT VIA MIDI

Described as the newest innovation in MIDI-controllable electronic patchbay technology, the EPP400 provides five effects loops (three in stereo) and 128 programmable preset locations controlling loop on/off and loop sequence.

Preset selection may be controlled by the front panel, an optional IFC60 Intelligent Foot Controller, or by input from any MIDI-equipped controller. The EPP400 can be assigned to any of the 16 MIDI channels for automatic patch selection.

CHESBRO MUSIC COMPANY

For additional information circle #223

MODEL X324 STEREO CROSSOVER FROM AUDIO LOGIC

The X324 can be utilized in either a stereo three-way or mono four-way capacity. Balanced inputs and outputs with phase inversion on all six outputs are provided, along with an additional summed output of low frequencies from both channels for mono sub-woofer connection.

Butterworth filters with 18 dB per octave slopes ensure driver protection by rapidly rolling off the frequencies at the crossover point, while the flat passband frequency response is said to eliminate peaks and dips in

the output. An electronic switch on the front panel adds a two-pole high-pass filter at 40 Hz.

No patching is required for mono operation since all connections are made internally.

Pro-user price of the X324 Stereo Crossover is $329.95

AUDIO LOGIC

NEW RACK CONSOLE FROM SKE

The Rack 'N Roller is a mobile equipment console designed for both studio and location applications, and allows recording and pro-

cessing gear to be housed into one compact package. A canted, six-piece effects rack is suspended over an angled recorder console. Preset electronics, such as noise reduction units, can be placed in the console's lower section.

The unit is available in both single- and double-rack configurations. The double-rack console, combined with optional Road Package, is said to be ideal for transporting two recorders in live multitrack and/or two-track, processed mixdown situations.

Custom-design include patch bay or power wiring; single-point grounding network; multipin snake system; and filler panels. Also available is the Road Package consisting of latched covers and heavy-duty locking casters.

The Rack 'N Roller single rack is priced at $425, and the double at $625.

S.K. ENTERPRISES

NEW XT:C DIGITAL
REVERB FROM ALESIS

Features include a 16 kHz frequency response, full stereo inut and output, and decay time variable from almost zero to 15

seconds, set by a single rotary control. The XT:c supports eight separate reverberation programs, each augmented by front-panel option switches. Selectable options include damping, LF cut, and infinite hold.

Rotary controls include Predelay (from zero to 200 milliseconds), decay time and high-frequency roll (16 to 3 kHz). Programs include a few small spaces, a fes large rooms, a huge hall and gated and reverse reverb.

Suggested list price of the XT:c is $749.

ALESIS

MIDI MAPPING UNIT
FROM AXXESS UNLIMITED

"The Mapper" is described as the first product to make complete use of MIDI with a new technology called MIDI Mapping, the ability to recognize, modify, and expand MIDI codes automatically. The Mapper is a stand-alone device that requires no more than a standard MIDI keyboard for programming. It can feed MIDI data back to the master keyboard, or intercept and alter commands on their way to other synthesizers.

In doing so, the unit can create any number of keyboard splits, which may overlap and be assigned to any combination of channels. Notes can be expanded ro play other notes. Controls (mod wheel, breath, etc.) may be routed to control multiple functions simultaneously, and are also channel assignable.

Performance set-up commands such as program changes, MIDI routing and exclusive messages, can be sent automatically.

All mapping functions may be stored in memory locations called MAPS (*Midi Altered Program Sets*), and recalled directly or stepped through with a foot pedal according to a sequence. The Mapper has two MIDI Ins and two Outs, and an RS-232 port for optional computer. List price is $975.

ACCESS UNLIMITED, INC.

www.americanradiohistory.com

TREBAS INSTITUTE OF RECORDING ARTS OPENS LOS ANGELES CAMPUS

The school currently has four campuses in Canada — Toronto, Montreal, Ottawa and Vancouver — and offers more than 80 courses in record producing, sound engineering and music industry management in a two-year, diploma program. Instructors include Tom Noonan, associate publisher and director of charts for *Billboard* (Communication); Joe Csida, former *Billboard* editor-in-chief and author of *Music Record Career Handbook* (Music Industry Overview); Gary Solt (Ear training and Music Theory); Scot Scheer (Record Producing); Myles Mangram (Management); and *R-e/p* writer Denis Degher (Sound Engineering).

The Institute is located at 6602 Sunset Boulevard, Los Angeles, CA 90028. (213) 467-6800.

JBL PROFESSIONAL TO HANDLE U.S. DISTRIBUTION FOR SOUNDCRAFT ELECTRONICS

The exclusive distribution of Soundcraft Electronics products in the U.S. and Mexico will now be handled by JBL Professional, according to Ron Means, president of JBL Professional, and Philip Dudderidge, chairman and managing director of Soundcraft Electronics Limited, England. The new division will be referred to as Soundcraft U.S.A.

"In an effort to strengthen Soundcraft's position in the U.S. and Mexican markets," Mean explains, "JBL Professional was appointed distributor because of their knowledge of marketing complimentary products to all areas of the professional sound industry."

Soundcraft U.S.A. will be a separate division within JBL Professional and, under the direction of Means, will continue to operate from its current Santa Monica, CA, offices. Continuing to head the sales and marketing effort for Soundcraft U.S.A. will be Wayne Freeman. Betty Bennett, president of Soundcraft in Santa Monica, will be retiring to raise a family, but will remain a consultant to Soundcraft U.S.A.

MASTER'S WORKSHOP TO INSTALL PAIR OF SONY PCM-3324 DIGITAL MULTITRACKS

Scheduled for installation in Master's recently renovated facilities, the twin system will make the studio complex the only Canadian organization capable of 48-track digital recording, along with 24-to-24 track electronic editing and dubbing.

Master's is described as one of Canada's leading studios in audio-for-video production. Recent clients include Disney, Imax, Omnimax, HBO, PBS, Showtime, CBC, CTV and Global, as well as live concert recordings for Supertramp, Police, David Bowie, and more.

TWICKENHAM FILM STUDIOS PLACES LARGE ORDER WITH QUAD-EIGHT/WESTREX

Phase #1 represents the delivery and installation of Westrex film recorders and projectors worth some $200,000. In Phase #2, the contract for which was awarded four months ago, Twickenham Studios will receive additional recorders and projectors to a value of $350,000, and scheduled for delivery next June.

The third phase of the order is a specification for a custom-built Quad-Eight console, complete with a Compumix IV automation system with Intelligent Digital Faders. This desk will be one of the central features in the new complex being built at Twickenham, scheduled for completion in August 1986.

The Twickenham console will feature 72 inputs, using Quad Eight's new input module specifically designed for film recording; 24 mix busses allow selection of any recorder from any operator position. Also included are 10 echo/effect send busses and independent three- and two-channel pan busses.

GOLD LINE TO MANUFACTURE LOFT PRODUCT LINE

Formerly produced by Phoenix Audio Labs, the Loft line includes the Model

cleaner final product.

Headroom

Console headroom is another topic that may need re-evaluation in the context of digital audio — just how *much* headroom is "enough" is a serious, sometimes emotional issue. We reminisce about those good old vacuum tube consoles that seemed to never run out of gas, yet without recalling their less than remarkable noise floors.

The soft overload/compression characteristic of analog recording tape makes the availability of extra headroom desirable; relatively small amounts of electrical clipping can sound harsh, even in the presence of numerically larger amounts of tape saturation induced distortion.

Since digital recorders behave very much like clipped electrical circuits during overload, excess headroom beyond this clipping point provides us with very little benefit. Hard clipping prior to the digital processing should be avoided, since such clipping generates excessive high-frequency energy that can cause ringing and overshoot in the anti-aliasing filters.

Most high performance audio circuitry runs from ±15 to ±22 volt power-supply rails. Modern low-noise integrated circuits, when operated from even ±15V supplies, can deliver dynamic range well in excess of any source or storage medium I know of. Arguments that larger than 30V peak-to-peak signal handling capabilities are needed in line level processing gear I find to be without merit.

I have come up with three exceptions where more than 30Vp-p can be useful. All three are output or interface related so they don't support the argument for additional level. The first and most obvious example, where larger swings are useful, is in the output of a power amplifier, where such a swing is necessary to develop adequate power in a speaker load.

A second example occurs in live PA snakes, where crosstalk and other induced-noise sources are often well above circuit noise floors. However, differential outputs (see the October 1985 *Audio Mythology* column) can deliver the equivalent of 60Vp-p from ±15V power supplies. The third case is for terminated, transformer-coupled distribution systems. A three-way constant impedance splitter transformer will be down 10 dB at each tap, in addition to the 6 dB insertion loss for a standard termination.

So far I have argued, with the above exceptions, that more than 30Vp-p is not really useful. This should not be interpreted to mean that any circuit using ±15V power supplies will have adequate headroom. There are several ways to run short of headroom, most notably in equalizer sections, fader/VCA make up

gain stages, and summing busses. Such areas can be designed and worked around, so long as you know where your headroom limitations exist. Small amounts of clipping will not automatically give you that "dead-battery" sound, just a slightly harsh, subtly inferior character.

Multiple peak indicators would be ideal, while a single indicator that samples several points in the signal chain would provide a good alternative. If you don't have, or don't trust, the indicators on your board, you can try injecting a simple tone, and listen for distortion. It is best to use a low-frequency (less than 500 Hz) signal, since the distortion products will then fall in the mid band where your ears are most sensitive. Adjust gains and listen at various patch points until you can determine where your limitations are. If you are one of the two or three people out there still using analog tape, you might find it interesting to hit the tape with a serious overload, and see how it sounds. (Overloading your digital recorder won't be nearly as interesting sounding.) ∎∎∎

References

1. *Performance Limits in Contemporary Console Design*, by John Roberts, *Re/p*, April 1980.
2. "Audio Mythology," April 1984.

INDUSTRY INTELLIGENCE

A Sneak Preview of the New Otari DTR-900 PD-Format Digital Multitrack

by John Carey, Marketing Manager, Otari Corporation

As previously announced at a joint press conference during the recent New York AES Convention, Otari Electric Company will exhibit its first digital multitrack at the International AES Convention in Montreaux, Switzerland, March 4 thru 7, 1986. The new DTR-900 is manufactured in accordance with the Professional Digital (PD) format standard also announced at the New York AES. The PD format agreement assures that tapes made on a PD machine will record and playback on any other PD machine. The Mitsubishi X-850, introduced at the last AES, is an example of one such compatible PD digital recorder.

The following sections describe the PD format's key technical points and, in particular, outline the features and capabilities of the Otari DTR-900.

Summary of PD Format

The PD format for fixed-head digital recording offers complete electrical and tape compatibility between PD machines, to assure the ease of backup and tape exchange between studios or stations. The format agreement includes 32 channels on one-inch tape, 16 channels on half-inch tape, and two channels on quarter-inch tape. The data-coding scheme used in the PD format yields digital recordings that are extremely resistant to errors caused by tape dropout or contamination of either the head or the tape surfaces. A combination of Reed-Solomon Coding (RSC), Cyclical Redundancy Check coding (CRC), and a high level of physical dispersion of the data across the tape media allows for both razor-blade splice and electronic editing without any loss of fidelity.

The 32-channel PD format consists of a total of 45 tracks recorded on one-inch tape. For each group of eight audio channels, two additional tracks of RSC parity information are generated for a total of 40 tracks (eight audio + two parity × 4 groups = 40). Five additional tracks are provided on the machine: two auxiliary digital, two auxiliary analog, and one timecode.

The data coding used in the PD format provides an additional benefit beyond its high level of error detection/-correction. Since output clocking of the data may be derived from any of the 32 audio data channels, there is no need for a separate control track.

Machine Description

Otari believes that it is necessary to provide the market with products that are capable of not only achieving, but maintaining, consistently high levels of performance during rigorous professional use. Since performance and reliability directly contribute to the success or failure of a particular user or facility, Otari has placed a special engineering emphasis on the DTR-900's long-term reliability.

Because of the considerable expectations audio professionals now place on digital recording, it has been our highest priority to design and manufacture an exceptionally reliable machine that will meet new challenges. With the addition of the PD digital recording products to our line, we will offer more different types of audio recording and duplicating products than any other manufacturer in the industry. Even within our digital product line, our plans call for product diversity.

The DTR-900 will be available in both 32-channel and 24/32-channel versions on one-inch tape. The 24/32 version will be similar in concept to the analog
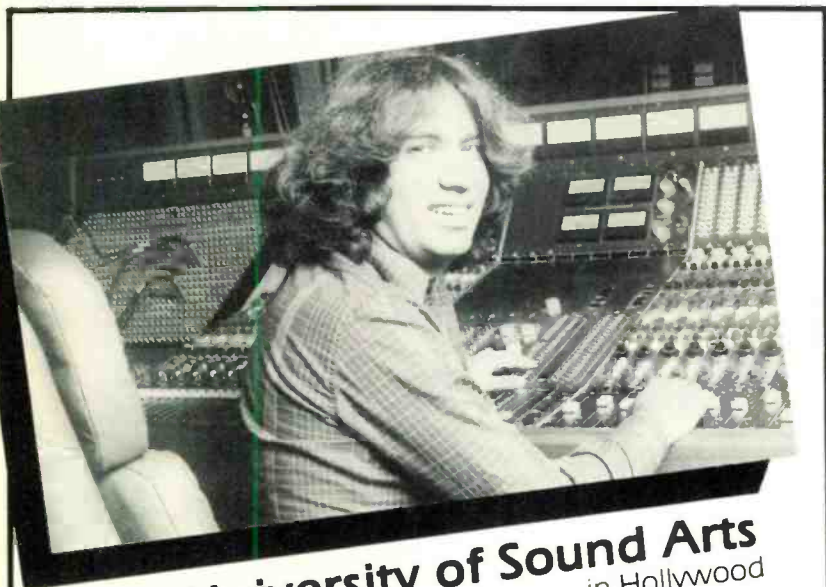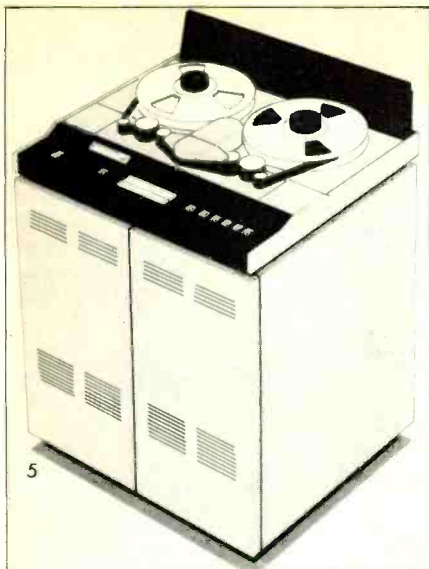
recorder models we currently offer wherein the user may, at a later date, easily expand the 16/24 or 8/16 machines to their larger channel configurations. The DTR-900-24/32 digital recorder will incorporate this same expansion concept, and will also offer another significant benefit. The DTR-900-23/32 will be supplied with 32 channel heads. In the 24-channel configuration, the recorder will be delivered with the capability to add the additional eight audio circuit boards (plus their associated dual parity electronics) at a later date. The advantages to this approach should be obvious in that previously recorded 24/32 tapes may be played and re-recorded on any 32-channel PD machine, and a 32 channel PD machine with all channels equipped can add an additional eight audio channels to the original 24 channel tape at a later date!

The DTR-900 Transport

The DTR-900 is based on the pinchroller-less transport developed and refined in the MTR-90 Series, the heart of which is a direct-drive capstan motor that governs tape motion in any operational mode, and provides smooth and precise bi-directional tape control from stop to 300 ips. When combined with the heavy-duty half-horsepower reel motors, the resultant tape handling characteristics are extremely stable and gentle. Tape tension is evenly distributed across the entire transport, from the supply reel to the take-up reel.

The DTR-900's surface is built on a two-inch thick, milled alloy casting for stability and durability. The transport will accept up to 14-inch reels, which permit as much as one hour of continuous recording.

Because the DTR-900 will be expected to easily synchronize to videotape or film systems, the excellent external machine control capabilities of the MTR-90 pinchroller-less transport can be anticipated. The deck will accept a wide range of external timebase references, including 50/60 Hz, 59.94 Hz, 9.6 kHz, 8 kHz, and composite video. Inter-

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face with machine control systems will be simplified because of Otari's standard 37-pin parallel connector. An optional plug-in timecode synchronizer will be available for the DTR-900 Series recorders; it will provide an RS-232C serial port for control of all synchronizing functions and, additionally, will support the SMPTE/EBU "ESBUS" protocol via an optional RS-422 port. The machine is capable of ±10% varispeed, a speed range that will be user-adjustable and displayed in 0.1% steps through the increment/decrement controls on the remote controller.

Audio Electronics

Analog inputs and outputs are active, balanced and operate at a nominal level of +4 dBm. The digital inputs and outputs are plug-compatible with all PD machines (e.g., the Mitsubishi X-800/X-850, or machines expected from AEG).

Audio performance of the digital recording and reproduce channels is consistent with current 16-bit PCM technology; 20 Hz to 20 kHz bandwidth, +0.5, –1.0 dB; over 90-dB dynamic range; inaudible wow and flutter; etc. The audio inputs will feature user-selectable pre-emphasis of 50 plus 15 microseconds. The A/D converters may be switched between 48 and 44.1 kHz sampling rates (the latter rate will not allow the use of the varispeed feature described above).

All 32 channels will have 20-segment LED meters for record and reproduce level monitoring, and an optional metering panel can be attached to the remote session controller. Audio record punch-in and -out is totally silent, seamless and gapless. An electronic crossfade time constant of either five or 10 milliseconds may be user-selected. When making an insert edit on one channel, only that channel and its two associated RSC parity tracks are recorded. The PD format's error-correction system permits mechanical razor-blade edits on the digital tape, without any detectable degradation in the playback signal. Because of the inherent flexibility in the format's encoding scheme, it is the only format capable of re-recording over a splice from a previously recorded tape.

The two analog audio channels are intended for recording reference information for use during razor-blade editing. The two auxiliary channels allow convenient tape storage of console automation, CD subcode, or MIDI data. A dedicated timecode channel is employed for tape-machine address location using the remote session controller or, alternately, for use by an external synchronizer, editor or machine controller.

Remote Session Controller

The DTR-900 Series machines will include a remote session controller that mounts on a variable profile stand, and includes all channel function controls,

as well as the autolocator and various special function controls. The remote section has all channels record ready/safe selects, output monitor status switching for tape or input, and includes channel control grouping with set-up memories for increased efficiency and accuracy in session control. Additional control features of the DTR-900 include: audio muting, input monitor standby switching, and variable speed control with an LED display. The autolocator section includes 99 memories for storage of timecode location when the tape has been pre-stripped with code.

The controller has all of the standard functions currently found on an Otari autolocator including: cue storage, cue search, search zero, and repeat. With on-board automatic record-event memories, the DTR-900 is capable of automatic record punch-in and -out. An insert edit and exit point may be rehearsed from the remote, and precisely trimmed before committing to the record mode. There is a user-definable pre-roll stored in memory for use as a preview/rehearse aid when selecting the ideal insert points. The tape-timing displays, which are included in the locator section, will display hours, minutes, seconds, and frames, with eight digits for additional ease of operation. Lastly, the remote duplicates the DTR-900's transport controls, and can be arranged in any order the user prefers.

Delivery of production models of the DTR-900-32 and 24/32 are scheduled for Summer, 1986. The machines will be available through a select group of professional audio dealers that currently distribute Otari MTR-90II machines.

Otari is planning to introduce the companion two-channel PD digital recorder — tentatively named the DTR-200 — by the end of 1986. ∎∎∎

KLARK-TEKNIK DN-780 REVIEW: PRICING UPDATES

Since the publication of Bob Hodas' in-use operational assessment of the DN-780 digital reverb and special effects processor, which appeared in the December 1985 issue of R-e/p, we have been informed of a price reduction for the unit, along with a new software upgrade and additional interface capabilities.

Effective immediately, the DN-780 has a recommended retail price of $3,900, including remote control, a drop of $1,400 from the unit's previous price of $5,500. According to Jack Kelly, president of Klark-Teknik Electronics, Inc., the price reduction results from a drop in parts' costs, specifically VLSI technology now available for the digital processor.

The new software EPROM, designated V1.6, will be available free from K-T dealers by March, and includes additional "enriched" chamber, plate and concert-hall programs.

Also scheduled to be made by April are two interface cards for the DN-780 to accommodate external MIDI and serial communication via RS-232. While final prices for the two interface boards has yet to be finalized, according to Kelly they will be made available at "modest cost" to current owners. ∎∎∎

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News

TS-1 Audio Test Set, a number of cross-overs, a parametric equalizer, quad gate/limiters, and a delayed auditory feedback processor for training.

Some of the former representatives of Phoenix Audio will also represent Gold Line. In most cases, however, Gold Line's present representatives will handle both product lines.

The company will also be setting up a repair facility at its New Bedford, MA, factory. Customers should contact Gold Line at (203) 938-2588 to obtain an RA Number and shipping instructions.

SPARS ORGANIZES SECOND ANNUAL BUSINESS CONFERENCE

To be held March 22 through 23 at the Graduate School of Management, University of California at Los Angeles, the forthcoming conference will be modeled on last year's successful gathering at 3M World Headquarters. The agenda will include seminars on the following six topics: Obedience Training for Bankers; Insurance Costs; Effective

Cash Control; Investment Analysis and Cash Flow; Management's General Business Forecast; and Developing Areas of Business for Studios. In addition, there will be a reception each evening featuring demonstrations of Studio Business Software by leading vendors in the field.

Registration fees for SPARS members will be $380 before March 1, and $425 after that date. For non-SPARS members, the fees will be $530 and $575, respectively. Participation will be limited to 70 registrants.

For more information, individuals and software vendors should contact the SPARS National Office, P.O. Box 11333, Beverly Hills, CA 90213. (213) 466-1244. ■■■

— News Notes —

• Effective December 1, 1985, **Phoenix Systems'** products will be manufactured under license and distributed by **Rhoades National Corporation**, Columbia, TN. A combination Rhoades/Phoenix catalog can be requested by writing: Rhoades National Corporation, Dept. R/P, P.O. Box 1316, Columbia, TN 38401. (615) 381-9007.

• **Sounds Good Audio**, Lansing, MI, has

BOOK REVIEW

Principles of Digital Audio, by Ken C. Pohlmann

Reviewed by Bruce Botnick and David Collins, Digital Magnetics, Inc.

As Ken Pohlmann says in the preface to his new book: "A book on digital audio? Is this really necessary? In today's problematic world of cruise missiles, chemical dump sites, acid rain, and fast food croissants, is a book on digital audio *really* important enough to be written?" You betcha, Red Rider! This book is our digital bible!

The fact that many people who use digital recording equipment on a daily basis have a limited understanding of the principals is proof of the extreme need of books of this type. Fortunately, Ken Pohlmann possesses the technical knowledge, combined with a wonderful flair for explanation and analogy (pun intended) that is required to make tech-

nical books readable. This should come as no surprise, however, since the author presumably has had a lot of experience in the area of digital technology in his role as an engineering professor at the University of Miami. Suberbly written and illustrated, *Principles of Digital Audio* allows the technical novice as well as the technical wizard the opportunity to understand the mysteries of digital audio.

It is quite an effort that Ken Pohlmann has put together. Simply put, the book covers eight fascinating chapters, starting with a discussion of the physics of sound transmission in air, and basic electricial concepts relating to sound reproduction in the analog domain.

Next, these familiar quantities are contrasted with their counterparts in digital information systems, such as sampling, quantization, signal-to-error ratio, aliasing, and dither. Ken's method of bringing you into the 21st Century is with such joy and enthusiasm that you can't help but get caught up in his excitement. He explains in great detail the entire Digital Audio chain from input to output.

First, a description of Pulse Code Modulation is in order, since this is the technique used by the majority of manufacturers today. Next, several systems of analog-to-digital conversion are explored — including successive approximation, dual-slope integration, parallel — with an eye towards their pros and cons in various audio applications. The concept of dither is introduced early in the book and is explained very thoroughly; it is certainly emphasized that the presence of dither can eradicate many quantization effects of PCM systems at the expense of a tiny bit of noise. Dither helps to make the digital system more "analog" in its operation, and many designs employ it in some form.

The all-important and much misunderstood lowpass filter is also studied in detail, with an emphasis on design criteria that maintain a flat passband and minimum phase impulse responses. These filters, whether digital or analog in nature, certainly have attained a hero/villian status in the sound of digital systems, and the coverage here is excellent and long overdue!

Next, of course, is the Digital-to-Analog conversion stage; in this chapter some different options are explored, with oversampling preceding D/A conversion being suggested as the most viable technique. Naturally, the ladder-type, integrating and dynamic element-matching converters are described and contrasted for audio use; other digitization methods, such as floating-point and adaptive delta modulation are also discussed.

The book provides a complete outline of the various storage devices available today for digital audio, including a discussion of longitudinal, perpendicular and isotropic magnetic recording techniques found in most commonly used stationary- and rotating-head recorders;

hard-disk computer systems; and an insight into the future storage possibilities offered by Optical Disc (DRAW, or Digital Read After Write) systems. The author also includes a look at direct satellite broadcast and cable digital audio/data transmission systems. Error detection, correction, and concealment strategies are explained.

An entire chapter is devoted to the Compact Disc. Will the CD replace the LP and cassette as the dominant consumer playback medium? Is it suitable for the automobile, and the beach?

In the book's last chapter, titled "A New Beginning," Ken addresses the possible end of analog processing; the promise of higher fidelity; and the sampling-rate controversy.

We recommend that you run, not walk, to your nearest technical book store [or send $21.95, including postage, to *R-e/p Books* at the address given below — *Editor*], and buy this book. If you are thinking of including digital technology as part of your studio, you should know all you can about what makes digital tick. Besides, Ken could use the extra money he will earn from sales of the book to buy a CD blaster to take to the beach! ∎∎∎

Principles of Digital Audio is available from *R-e/p Books* for $21.95, including postage. When ordering, please send a check or money order payable in U.S. funds to the following address:

R-e/p Books, P.O. Box 2449,
Hollywood, CA 90078

ordered a 52-position, 40-input version of **Harrison Systems'** new HM-4 stereo sound-reinforcement console system. The HM-4 will form the heart of SGA's main touring system that will travel first with Kashif on a 15-week tour scheduled to begin in mid-January.

• **Audio Engineering Associates**, Pasadena, CA, has been appointed the exclusive dealer for all **AMEK** series consoles in Southern California, according to Bob Owsinski, VP of sales and marketing for AMEK Consoles, Inc. "We are proud to be associated with a company that has such a fine technical reputation as well as a high degree of credibility in the Southern California audio community," Owinski states. "AEA has always been noted for selling only the best sounding, highest quality products, and in this regard we feel that they are the perfect company to represent us in this area. Also announced was AEA's appointment as a dealer for all Total Audio Concepts (TAC) products.

• **Audiotechniques** has delivered 50 **Sony** JH-110C Series tape machines to the U.S. Army. According to company president Bob Berliner this latest order represents the last production run of JH-110Cs to be manufactured by Sony's Fort Lauderdale facility. The company has also delivered two **Compusonics DSP-2002** digital editing systems to **Transcom Media**, New York City, for all audio post-production on a series of cartoons the facility is producing for network TV. The DSP systems incorporated a number of recently added features, including Automatic Dialog Replacement, Sony PCM-1610/F1 interface, video-sync reference, and butt splicing. ∎∎∎

You rehearsed long hours. Spent more hours laying down tracks. And a few more adding overdubs. Now it's time to mix down to your stereo master. This is no place for compromise. Insist on a two-track recorder from Studer Revox, a company dedicated to music mastering perfection.

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Like its "big brothers" in top studios, the PR99 MKII is built for long-term performance. The transport chassis and headblock are solid diecast aluminum, milled and drilled with exacting precision. So the parts fit together right. And stay together for a long time.

The PR99 MKII is also loaded with professional features: Balanced and floating +4 inputs and outputs. 10½" reel capacity. Tape dump. Self-sync. Input switching for tape echo effects. Output mode switching. Edit mode switch. And built-in front panel vari-speed. Options include steel roll-around console, monitor panel, and remote control.

The new microprocessor-controlled counter / locator saves time (and cools tempers!) in tricky mixes. Touch a button and go to zero. Exactly to zero. Touch another button and go to a locate point, which can be entered from the keyboard or "on the fly" from the counter. Because PR99 MKII finds your cues, you can concentrate on your mixing.

In overall sound quality, we believe the PR99 MKII once again steps out in front of the competition. So when you finish mixing, you hear a playback which captures all the excitement of your tracks.
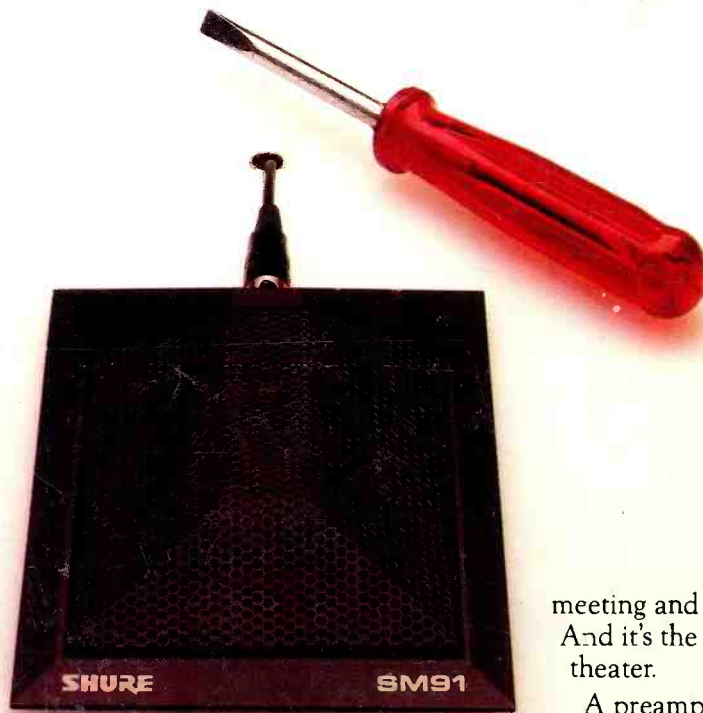
But that's for your ears to decide. Contact your Revox Professional Products Dealer and arrange an audition. Why settle for less than mastering perfection?

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