RECORDING ENGINEER/PRODUCER

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

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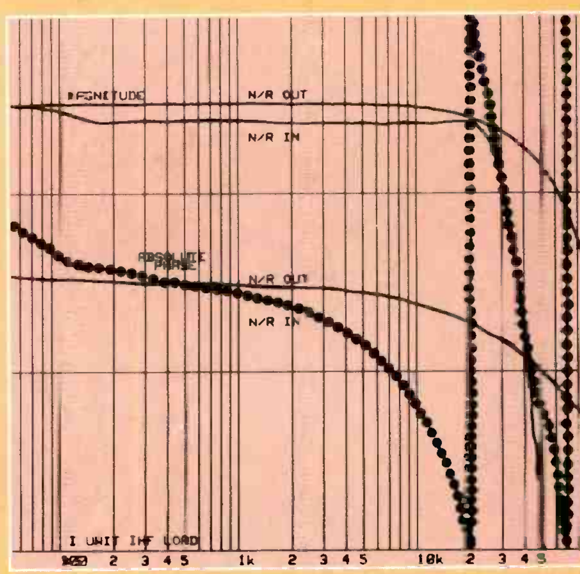
April 1985
Volume 16 — Number 2

EQUIPMENT ASSESSMENTS:
Tascam MS-16 — page 168
Eventide SP2016 — page 178

Producer JAY GRAYDON — Interviewed on page 34

SOUND CONTRACTING — page 104



EQUIPMENT REVIEW — page 168

Stationary

From 2 to 48-channel, the industry is going our way.

Tomorrow's recording studios will all be digital. In fact, the digital revolution is well under way. So what's the best way to get started now?

With Dash, you know you'll be ready for whatever the digital future may bring. Dash has the technical advances and the compatibility you're looking for. That's why so many more of the major manufacturers are backing it.

Dash offers up to 48 channels to meet the most demanding recording requirements, and because so many studios are turning to Dash, you'll have the confidence that your masters conform to the industry standard format. And that's a comforting fact.

And no one knows Dash like the people who started the whole digital revolution in the first place — Sony.

We've got the PCM-3324 multi-channel recorder ready to go right now and soon we'll be offering the PCM-3102 2-channel recorder.

In fact we've got more than 100 PCM-3324s already in the market.

When you choose Sony Dash multi-channel recorders and digital audio equipment, you're making a sound investment in the digital future.

We're in the Dash all the way.

DASH Digital Audio Stationary Head

SONY®

Head

For additional information circle #1

RECORDING ENGINEER/PRODUCER

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

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

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April 1985
Volume 16 — Number 2

— Contents —

1975: THE YEAR OF 24-TRACK ANALOG

1985: THE YEAR OF 32-TRACK DIGITAL

10 YEARS AHEAD OR BEHIND?

Remember 1975? The major studios were scrambling to update to 24-track.

One thousand 24-track machines and 10 years later, the world class studios have chosen the next generation: The Mitsubishi 32-Track Digital Audio Recorder.

The only multitrack with the true competitive edge.

THE X-800
by MITSUBISHI

News Letters Views

MUSIC FOR FILM

from: **David "Doc" Goldstein**
Goldwyn Sound Facility
Warner Hollywood Studios,
Los Angeles

I just finished reading your interview with Joe Chiccarelli [February 1985 issue], which I enjoyed very much. Not only do I enjoy and respect his work, but there were some good engineering ideas in the article. I am, however, concerned about one aspect.

When Joe was asked about what the production differences were when doing music for film, he talked about going for the feel and the attitude. Well, I agree. When a movie is mixed to the Academy Mono format, "feel and attitude" is all you *can* go for, because the high-end is drastically rolled off, starting at about 2 kHz.

Anything you do that is subtle will never get past that type of monitor curve. *However*, when the film is to be mixed in the Dolby Stereo format, everything changes. Now the monitor curve is wide range, and things can be heard clearly — but this can cause a different set of problems.

First of all, the Dolby [4-2-4] matrix does some very funny things to stereo record-type mixes. Things that are in phase across two stereo tracks (like stereo miked piano), and which fill a wide spread on a typical record mix, will be pushed into the center channel by the Dolby matrix, and suddenly you have *mono*. If you have done this mix with a careful eye to correct phase, chances are that the *whole* mix will sound mono. Also, things that should be out front in the mix like vocals and guitar solos, will tend to back up in the mix and be "backscreen."

Of course, the best way to avoid this is to mix the music in a dubbing stage instead of a control room, but that might not always be possible. Alternatives include mixing in a record studio, but using a Dolby matrix unit installed into the monitor system, or mixing to one-inch eight-track, using separate tracks for instruments: left, center, and right, and then left, center, right vocals and solos, plus edit code or sync tone on empty tracks — or, better yet, combining both of the above methods. That is what the Bee Gees did for *Staying Alive*, and it worked out very well. In this way the re-recording mixer can better control the music in the dub, and keep tracks that want to be apart separated by using digital delay lines or whatever may be necessary.

This letter is not really for Joe; after hearing his work, I doubt that he can learn anything from me. But others who read the interview may be able to benefit from this discussion. If you as an engineer are about to do a mix for a film project, I strongly suggest that you do a little checking. Contact Dolby. Talk to whomever the re-recording mixer will be. Come down to the dubbing room so that you will know how different the formats can be. I can't even count how many times people have come down with their two-track mixes, and were totally unprepared for how different it would sound in a film format.

When the Dolby matrix takes four channels (left, center, right, and surround) and encodes them to two channels for the optical Lt-Rt mix and then in the theatre decodes them back to four channels, *strange* things can happen if you aren't prepared. □□□

Editors Note: A full discussion of the Dolby Stereo process for film production, written by our film consulting editor, Larry Blake, will be published in the June issue, and will contain a technical sidebar detailing the increasing use of rock music in films, and how best to accommodate the points raised in Doc Goldstein's letter.

MICROPHONE ASSESSMENTS

from: **John Oster**
Classic Recording
Sacramento, CA

I have found Lowell Cross' articles on "Performance Assessments of Studio Microphones" [April, December '84 and February '85 issues] to be very informative, but also a bit frustrating. As he pointed out, some of his personal bias entered into the assessments, but I don't feel he gave the B&K 4007 a fair shake. No engineer in his right mind would space them nine feet apart to record a soprano and piano! About three or four feet would be plenty.

I own a pair of B&K 4006s and have used them many times, and have achieved very satisfactory imaging. I admit [that] in certain circumstances, coincident placement might be better but, overall, spaced omnis have a much more pleasing tonal balance and spaciousness. A large part of the soundstage is off-mike with coincident techniques (except M-S) and this does color the sound.

I record about 60 to 70 concerts each year for broadcast on an NPR station and I do resort to coincident miking for mono compatibility. But when *sound quality* is of paramount importance, I use spaced omnis.

Mr. Cross is in an enviable position to assess so many good mikes, and I hope that he continues as there are many new models coming on the market. I just hope he will continue to be more open minded with omnidirectional microphones.

Lowell Cross replies:

Omnidirectional stereo recording and the qualitative properties of individual microphones are two different issues. As stated in the article, the spacing of the 4007 omnidirectional microphones was *reduced* to 2.8 meters (nine feet) from the four to five meter (13 to 16 feet) distance recommended on two occasions by the B&K representative, Henning Moller. Not only do I agree with John Oster that even this reduced spacing is too wide, but so does the stereo correlation meter. I advanced the opinion that the 4007 almost certainly would have received a higher composite "rating" from our group of listeners if it had been represented by a single microphone, placed *closer to the singer*.

Mr. Oster suggests three to four feet as an acceptable spacing for recording in stereo with omnidirectional microphones. In our experiment, the PZM™ microphones (with hemispherical polar patterns) were spaced exactly four feet apart, resulting in unsatisfactory correlation readings of -0.4 to +0.6. Furthermore, a noticably diffuse and amorphous stereo image was obtained from this set up when directly compared to the near-coincident, artificial head, MS, and XY arrangements. So I maintain that three to four feet is *also* too great a spacing for preserving a coherent stereo image, one that is free from exaggerated random phase differences between channels. Of course, bringing omnidirectional microphones into a closer proximity to each other for stereo recording reduces channel separation eventually to zero. One is simply faced with making compromises when attempting to record in stereo with omnidirectional microphones, with the exception of the use of an artificial head.

A careful reading of the article will reveal that the B&K microphones *as individual units* received very complimentary remarks indeed from the author — certainly a "fair shake." They were acclaimed with highest praise as "truly excellent" and "state of the art" microphones along with the Neumann TLM170 and U89, Schoeps MSTC54 and CMC54U, Calrec MkIV Soundfield, and Neumann KU81. Special mention was made of the qualities of the 4006 and 4007 units when used for solo (i.e., monaural) pickups, in which applications they excel. I can only repeat my suggestion that B&K should consider marketing the 4000 series of pressure transducers in an artificial head so that wide spacing (even three to four feet) is not required for their use in stereo applications. . . . continued overleaf —

Letters

Incidently, some typesetting errors, which might have caused a certain degree of confusion, crept into this article. To clarify matters, the corrections are:

- On page 74, a paragraph which began "As noted earlier, Bruel and Kjaer . . . should have read "As noted earlier, Bruel and Kjaer is committed exclusively to the manufacturing of omnidirectional microphones for audio recording";
- More seriously, on page 80 we inadvertantly identified "Four Hypercardioid Ambisonic Surround Sound Pattern" on the left-hand drawing when, in actuality, the caption should have read "Four Cardioids"; in essence, the two drawings were reversed. ■■■

EQUIPMENT LIFE

from: **Fred Krock, engineering supervisor**
KQED-FM
San Francisco, CA

Congratulations to John Roberts for two years of "Exposing Audio Mythology." I have enjoyed every one of the articles; his column fills a real need.

For one answer to the question of whether it is better to leave equipment on or turn it off, I refer you to the *Federal Power Tube Handbook.* (Federal was a respected manufacturer of transmitting tubes for many years.)

Federal made some extensive studies of this question in the late Thirties. Their conclusion was that the break-even point for transmitting tubes is four hours. If the equipment is not used for more than four hours, it should be turned off; less than four hours, leave it on.

The validity of this conclusion for audio equipment can be questioned on these grounds:

1. The study was based on tube life, and did not include any other components;
2. The study was based on bright tungsten filament transmitting tubes, not small, indirectly heated cathode receiving-type tubes similar to those used in audio; and
3. The cost of electricity to illuminate filaments and operate cooling blowers was included in the study. Some transmitting tubes use a lot of electricity for these purposes. However, the cost of the electricity probably is offset by the higher original cost of the transmitting tube compared with the receiving tube.

My own observation over many years is that receiving-type tubes in a transmitter operating 18 hours a day have similar life times to the same tubes in monitors and audio amplifiers operating 24 hours a day.

I have used the four-hour rule for professional tube-type equipment for many years. It's probably as good as any. However, be advised that Murphy's Law has a special corrolary stating that when power switch will fail if it is used regularly. □□□

News

SEMINAR ON IMPROVING AUDIO CASSETTE QUALITY

Electro Sound, Inc, has organized a seminar entitled "Applications for Better Cassette Quality," to be held at the San Francisco Hyatt on Union Square Hotel, from August 20 thru 23. Co-sponsored by sèveral record companies, manufacturers and suppliers of raw materials and duplicating equipment — including Agfa-Gevaert, Athenia, BASF, Capitol Magnetics, Columbia Magnetic Tape, Mitsubishi/DEC, Dolby Laboratories, DuPont, Hercules, ICM, Ltd., IPS, Inc., Saki Magnetics, Shape, Inc. and Studer/Revox America — the seminar will cover the various parameters that artists and producers expect from pre-recorded cassettes; standards and quality-control measures; raw-materials advances; duplication equipment alignment; and where (how) do we go from here?

According to Electro Sound/Sunnyvale president Robert Barone, the primary focus of the 1985 seminar will be "to examine what we are trying to accomplish, discuss how we do it now, and define what we need to make a better quality cassette."

More details from Electro Sound at: (408) 245-6600.

LOS ANGELES RECORD PLANT TO RELOCATE FROM THIRD STREET

After 16 years at its original Third Street location in West Hollywood, in September the Record Plant's recording facilities and offices are scheduled to move to the former Radio Recorders Annex on Sycamore Street, Hollywood. Choice of the new site —in the heart of the city's film and video post-production area — is said to reflect the company's growing involvement in the field of audio for motion pictures and video. In announcing the move, however, studio president Chris Stone stressed that the studio "would continue to service its major record-industry clients in the mannner to which they have become accustomed. While our daytime schedules will be primarily film and TV work, the nights will continue to belong to record projects. Our involvement in records *and* the visual media allows us to promote . . . better music quality for both industries."

Recording facilities at the new location will consist of two large scoring stages, plus a video scoring/record overdub and mix room. The scoring stages — one of which measures 3,500 and the other just under 2,000 square feet — will have full mag and projection capabilities. The 1,350 square feet video-scoring room will also serve as a showroom for Audio Intervisual Design, Record Plant's pro-audio sales/consulting division.

Along with the studios, the Sycamore Street location will house all of the Record Plant's satellite companies, including Livingstone Audio rental and Digital Electronic Leasing, plus Studio M, Inc. and Record Plant Scoring.

The move is planned to take place in stages during September, so that there will be no time lapse between the closing of the old location and opening of the Sycamore Street site.

WORLD'S FIRST "MUSIC VIDEO POSTCARD" RECORDED DIGITALLY WITH MITSUBISHI X-800/X-80

Criteria Studio, Miami, recently was involved in the recording and post production of what has been described as the world's first "Music Video Postcard" — a completely digital recording for release soley on high quality VHS and Beta HiFi formats — for Nicholas Communcations, a new video software company based in Washington, D.C.

The video release takes the form of a guided tour to the sight and sounds of the nation's capital, and incorporates sound effects recorded entirely on-location with an Audio+Design/Calrec Soundfield microphone. At Criteria the various effects, dialog and digital mulitrack music elements — which also were recorded using the Soundfield microphone system — were combined via an Audio+Design/Calrec Ambisonic Mastering System to produce a two-channel UHJ mix which, with suitable decoders, will provide listeners with what is described as a realistic surround-sound playback. Even conventional stereophonic replay of the UHJ-encoded soundtrack material is said to provide a sense of depth and realism not possible with normal techniques. All multitrack recording and mastering was done at Criteria using the facility's Mitsubishi X-800 digital 32-track and X-80 two-track machines.

According to executive producer Stephen Nicholas, "We are trying to create the finest piece of audio product ever made —one that bathes the viewer in the sense of being in Washington. The combination of visual imagery that we have selected, and the imagery that we were able to produce with the digital recorders is simply breathtaking. Our main goal with this project was to deliver to the consmer the finest piece of product that I can possible make, and one that completely envelopes the viewer in sights and sounds."

- **ALPHA AUDIO ACOUSTICS**, a division of Alpha Recording Corporation, has opened its third distribution center for Sonex acoustic foam. The new warehouse, located in North Las Vegas,

. . . continued on page 14 —

Turn this knob and enter your new recording environment.

Master-quality the Syncla

Synclavier is the world's most comprehensive digital music system. With its industry-leading new features, it offers the musician sounds of dazzling realism, along with extensive programmable control and wide-ranging facilities for composing, recording, editing, and performing.

Best of all, you don't have to be a technical whiz to use the Synclavier! Designed for musicians, the system is easy to learn and now includes special features which increase its musicality.

If you're thinking of building or expanding your studio, the Synclavier Digital Music System is a must. The Synclavier has been proven time and time again by top name artists and studios to be a tremendously creative and cost savings piece of equipment. In the best tradition of New England Digital, the following new Synclavier options prove once again why the Synclavier continues to lead in technological development.

Polyphonic Sampling

(16-Bit/100 kHz)

Concert grand piano, rich string sections, sizzling brass and the ultimate drum timbres are just a few of the unbelievable possibilities with the Synclavier's new Polyphonic Sampling Option.

Believe us, these timbres don't sound like you've got cotton in your ears. The full dynamic range rings true. This capability is provided by offering full 16-bit resolution with a user-variable sampling frequency up to 100 kHz. Expandable from 8 to 32 fully polyphonic voices. In addition you can order up to 32 — that's right — 32 megabytes of sound sampling memory (in 1 megabyte boards)! All voices are stereo and offer 96 dB S/N ratio. Plus, all voices can be controlled from the Synclavier's front panel in real time. A library of polyphonic sampled timbres featuring grand piano, strings, brass, and percussion is provided.

Sounds hot! Believe us, and your ears, it is!

Multi-Channel Independent Outputs

Once you have that finished recording in your Synclavier, you can now very easily link the Synclavier recorder to any multi-track recorder using the new independent output option.

The Multi-Channel Output Option allows you to route each track of the Synclavier's 32-track Digital Memory Recorder to a selectable individual output channel. Each output channel can be equalized or processed to produce a 32-track master, as well as a standard stereo composite.

This option may be expanded from 8 to 32 individual outputs as your Synclavier or recording capabilities expand. The option works with the regular FM synthesizer voices as well as the new polyphonic sampling voices.

76-Note Velocity/ Pressure Keyboard

Designed for musicians, the Synclavier 76 note programmable velocity/pressure keyboard provides quick and easy access to all the different features of the system, such as: a 32-track Digital

New England Digital Authorized Distributors **Atlanta** Songbird Studios 404/351-5955 **Boston** Syntone Inc. 617/267-4137 **Dallas** Lightning Music & Sound Inc. 214/387-1198 **Israel** Syntron

recordings from vier® studio.

Memory Recorder (which functions similarly to a multi-track tape recorder), plus independent programmable velocity and pressure, over 500 sounds instantly available, programmable split keyboard, pitch and modulation wheels, breath and ribbon controller, 192 different patching capabilities, plus much more!

Automated Music Printing

The first of its kind and still the leader has advanced another step with the refinement of additional software features. Complete scores, individual parts, piano scores and lead sheets are easily reproduced with incredible results.

An editor is included which allows you to perform typing in of lyrics, changes to the actual music, or adding of commands which will produce correctly transcribed triplets, quintuplets, and other irregular rhythmic groupings.

SMPTE

Pop your video monitor on top of your keyboard, lock up your Synclavier and video machine using the new SMPTE reader option and — presto! — compose the score with master-quality sounds and your music product is finished.

The SMPTE option allows you to position the master tape to any point. When the tape starts, the Synclavier will chase to the correct position. This avoids having to start the Synclavier and tape back to zero for each take.

The Option consists of the Reader/Interface Board, and special software. The reader unit handles 24 FPS (Film), 25 FPS (European), 30 FPS (Video), and Drop Frame Mode (Color).

MIDI

Of course we're doing MIDI. MIDI will be available in June as a simple retrofit to any Synclavier system.

Improved "User-Friendly" Software

In order to facilitate the operation of the system, New England Digital's software engineers have been busy restructuring the software interface to be very user-friendly. In addition, the system will feature a rhythm input page software routine for quick and precise entering of rhythm parts. Plus, an easy-to-follow set of menus which guide the user through any part of the system quickly.

Instructional Video Cassettes

If you're interested in relaxing at home and learning the basics of the Synclavier system, you can now purchase three video cassettes which guide the viewer through its basic features and operations. Send your check for $175.00 per set (not sold separately) plus postage and handling. Complete printed documentation is also available for $200 per set.

For more information or a personal demonstration, please call New England Digital or one of our authorized distributors:

New England Digital — White River Jct. VT 802/295-5800
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News

— coninued from page 10 . . .

Nevada, will serve West Coast accounts, and augments the existing facilities in Richmond, Virginia, and St. Paul, Minnesota. The most frequently ordered sizes and colors of Sonex acoustic-control materials will be stored at the new location, the company says, in quantities suitable for average sized installations.

• **HARRISON SYSTEMS** is providing a 24-input PRO-7 console for the U.S. Pavillion at the Expo '85 Exhibition to be held this summer in Tokyo. The mixer will be used in a computer-music presentation that has been prepared by the University of Buffalo under the auspices of the U.S. Information Service.

• **SANYO**'s Compact Disc Pressing Division has released the first commercial CD to utilize the company's High Reliance technology. The release, Mobile Fidelity's *Woodstock*, is a four-disk limited edition set, and described as the largest non-classical release to date, with each copy individually numbered. Compared to a conventional CD, a High Reliance Compact Disc is said to be less vulnerable to damage caused by exposure to high temperatures, and has a more durable surface coating to protect it from dust and scratches. Given their enhanced durability, HR CDs are claimed to be ideal for use in cars and portable players, as well as for audiophile and pure-digital releases, institutional uses and for CD ROM data storage.

• **THE ASSOCIATION OF PROFESSIONAL RECORDING STUDIOS**, London, England, has been evaluating a new interactive Viewdata service that will show available session time at member studios and cutting facilities. The new service, to be called *Studio Link-up*, will be made available via the Prestel system from British Telecom — the U.K. equivalent of ITT Dialcom, Delphi, CompuServe and similar data communications networks. Day-to-day operation of the proposed service will be handled by Gwynn Williams Viewdata Limited, a company that already operates similar services known as Theater Link-up (for venue availability and bookings), and Conference Link-up. Subscribers that access the APRS Prestel "pages" will be able to make immediate booking enquiries after checking available dates and times of uncommitted sessions at member studios. Also available will be an APRS Notice board to allow user-group members to place messages of general or specific interest. It is estimated that the U.K.'s top 15 record companies already have access to the Prestel Service — in order to receive the BPI's weekly charts — and that at least 95% of the country can access Prestel via a local phone call.

• **RUPERT NEVE, INC.** has received an order for a Digital Sound Processor (DSP) console from the Westdeutscher Rundfunk radio network, West Germany, for installation at the new Cologne Philharmonic Hall in the second quarter of 1986. Representing the sixth all-digital console to be supplied by the company, the new board will be configured for radio broadcast applications. Other DSP consoles, which because of their complexity can take up to one year to build, have been supplied to the British Broadcasting Corporation; CTS Music Center, London; Tape One Studios, London (a Compact Disc mastering desk, plus a digital disk-mastering console); and the National Archives, London.

NILE RODGERS, who has produced albums for Duran Duran, Peter Gabriel, Madonna, David Bowie, Sister Sledge, Kim Carnes and Mick Jagger, is said to have shifted his production methods from analog to recording on a Sony PCM-3324 DASH-format digital mulitrack. Currently operating from his home base at Power Station studio, New York, Rodgers' introduction to digital technology occurred during the producion of two Peter Gabriel tracks for *Gremlins*. "I had been working intensely for two weeks on the Gabriel tunes," the producer recalls, "and digital had created a standard of listening. When I returned to the analog project, I had the same engineer but couldn't understand why it suddenly sounded so different. I had grown accustomed to lack of tape hiss, for example. Digital also has a very solid bass response, a fuller high-end, and there are considerably more transients. I kept looking at the engineer and wondering what was wrong! It wasa learning experience."

Following the Gabriel session, Rodgers purchased a PCM-3324, which is now kept at Power Station for his own projects and for other studio clients. For multiple-machine sessions, he books time at Atlantic Studios in New York, where two additional -3324s are available. Rodgers' latest Duran Duran production made use of dual synchronized digital multitracks for a project that started in England with analog basics.

MORE NEWS on page 198
"Travels With the Editor" begins on page 25

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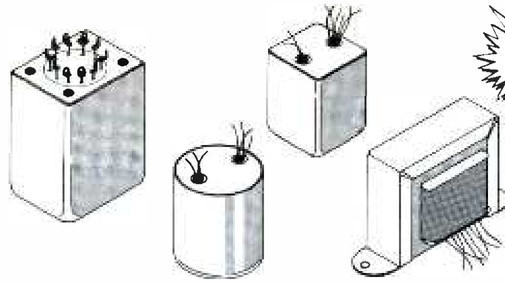

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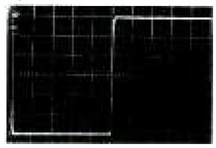


JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width[2] −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	100-249	1000
MICROPHONE INPUT																
† JE-16-A JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	−0.08/−0.05	200	−8	<1	1.7	−30	1	A=1 B=2	64.21 68.86	42.89 45.99	29.60 31.74
† JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	−0.09/−0.21	85	−19	<2	2.3	−30	1	A=1 B=2	64.21 68.86	42.89 45.99	29.60 31.74
JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	−6	0.170/0.010	−0.50/+0.10	115	−5	<7	1.5	−30	1	3	42.03	28.07	21.92
LINE INPUT																
JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	−0.03/−0.30	52	−28	<3		−30	1	1	103.47	69.13	47.69
JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	−0.03/−0.25	85	−23	<1		−30	1	3	40.05	26.76	20.90
† JE-6110K-B JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	−0.02/−0.09	125	−12	<1		−30	1	B=1 BB=2	62.86 71.52	42.01 47.79	30.83 32.97
* JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	−0.11/−0.08	160	−9	<2		−30	1	3	41.56	27.76	19.16
JE-11SSP-8M	Line in/ repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	−0.03/−0.00	120	−9	<3.5		−30	1	4	151.90	101.47	70.01
JE-11SSP-6M	Line in/ repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	−0.25/−0.00	160	−5	<3		−30	1	5	79.22	52.91	36.51
SPECIAL TYPES																
† JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	−0.16/−0.13	100	−12	<1		−30	2	3	34.60	23.13	18.06
† JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	−0.14/−0.16	100	−12	<1		−30	3	3	60.09	40.15	31.35
JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	−0.10/−1.00	40	−18	<1		−30	4	1	96.90	64.73	44.66
JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	−0.20/−0.20	80	−18	<1		−30	2	6	43.57	29.11	22.73

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W	L	H
1 =	1 5/16" Diam.		× 1 9/16"
2 =	1 3/16"	× 1 3/16"	× 1 5/8"
3 =	1 1/8" Diam.		× 1 1/16"
4 =	1 1/2"	× 1 3/4"	2 1/2" w/solder terminals
5 =	1 5/8" Diam.		× 1 3/4"
6 =	1 1/8" Diam.		× 15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot[8] (%)	Package[9]	PRICES 1-19	100-249	1000
* JE-123-BMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+28	2	−1.1	20Ω	0.002/0.002	−0.02/−0.02	>450 160	−2.1 −4.1	<1	7	87.41	44.17	30.47
* JE-123-DMCF	Quadfilar 80% nickel	600-600 150-600	1:1 1:2	+21	2	−1.0	19Ω	0.004/0.002	−0.02/−0.00	>450 230	−1.2 −2.5	<1	8	50.71	33.88	23.38
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	−1.1	20Ω	0.041/0.003	−0.02/−0.01	>450 170	−1.9 −4.0	<1	7	61.30	35.79	24.70
* JE-123-DLCF	Quadfilar	600-600 150-600	1:1 1:2	+27	2	−1.0	19Ω	0.065/0.003	−0.02/−0.01	>450 245	−1.2 −2.5	<1	8	39.61	26.45	19.42
JE-123-SLCF	Quadfilar	600-600 150-600	1:1 1:2	+23.5	2	−1.1	20Ω	0.088/0.003	−0.03/−0.01	>450 245	−1.2 −2.8	<1	9	33.48	22.35	15.43
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	−1.6	29Ω	0.114/0.003	−0.03/−0.01	>450 205	−1.2 −3.2	<1	10	25.48	17.01	12.49
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	−1.3	8Ω	0.125/0.003	−0.04/+0.06	190	4.6	<6	8	42.14	28.15	19.42
JE-11S-LCF	Bifilar w/ split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	−1.7	63Ω	0.058/0.002	−0.02/+0.01 −0.02/−0.05	>10MHz 155	+1.1 −4.1	<1	8	42.14	28.15	19.42

6. Multifilar construction has no faraday shield: cannot be used as input transformer. All specifications are for 0Ω source, 600Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier −3 dB @ 100 kHz
9. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available.

† IMPROVED PERFORMANCE * NEW MODELS

PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7 =	1 1/2"	× 2 5/16"	× 1 15/16"	2 13/16"
8 =	1 5/16"	× 1 15/16"	× 1 5/8"	2 3/8"
9 =	1 1/8"	× 1 11/16"	× 1 3/8"	2"
10 =	1 1/16"	× 1 7/16"	× 1 3/16"	1 3/4"

Prices shown are effective 6/1/84 and are subject to change without notice. Packing, shipping, and applicable sales taxes additional.

These charts include the most popular types which are usually available from stock. Many other types are available from stock or custom designs for OEM orders of 100 pieces or more can be made to order. Certified computer testing is available for OEM orders. Call or write for applications assistance and/or detailed data sheets on individual models.

jensen transformers
INCORPORATED

For additional information circle #7

The TS24 is the first in-line console from Soundcraft. And it represents a major breakthrough in in-line technology, because it now makes the console far easier to understand and operate.

Believe us, this is no hollow promise. Our argument is built around two rock solid foundations. Firstly, a new concept in console layout so logical, engineers used to split or in-line consoles can start work from day one. And secondly, a set of master conditions so advanced they'll amaze you.

STATUS.

One touch of the status button will configure the whole console for each particular stage of recording, mixing, broadcasting and video post production without sacrificing any flexibility whatsoever. In other words, one touch and you're off and running.

NEW DESIGN.

Conventional in-line consoles suffer from the limitations of one long travel fader and one equaliser being shared by two signal paths. With the engineer fader reversing and moving the equaliser back and forth throughout the recording, overdubbing and mixing process to optimise the situation.

The TS24 eliminates these shortcomings, thanks to its logical design. The long travel fader is in the section called MIX, which is the signal path for both monitoring and mixing. The equaliser moves between the MIX and CHANNEL signal paths automatically by use of the master status switches. 'Soft' switches may locally move EQ and AUX sends between the two signal paths but are also automatically reset.

When mixing, the Channel sections become available as additional inputs or effects sends without the limitations imposed by more conventional designs.

DROP-IN. BOUNCE.

Drop-ins are made easy by the use of the TAPE and GROUP button (T & G). Tape and Group enables you and the musician to monitor the original track and the overdub simultaneously.

The Bounce button facility enables you to take any combination of channels with their fader and pan settings directly to the routing matrix giving you instant bounce down.

SOUND AND VISION.

To create perfect sound, you also need perfect vision. With the TS24, that's exactly what you get. Separate scribble strips are provided instead of the usual confusing double one, and the Mix and Channel controls are in clearly defined areas for easier use.

AUTOMATION.

Soundcraft have developed a unique interface to the disc based MASTER MIX automation system, which enhances its operational flexibility by totally integrating the full extent of the console muting.

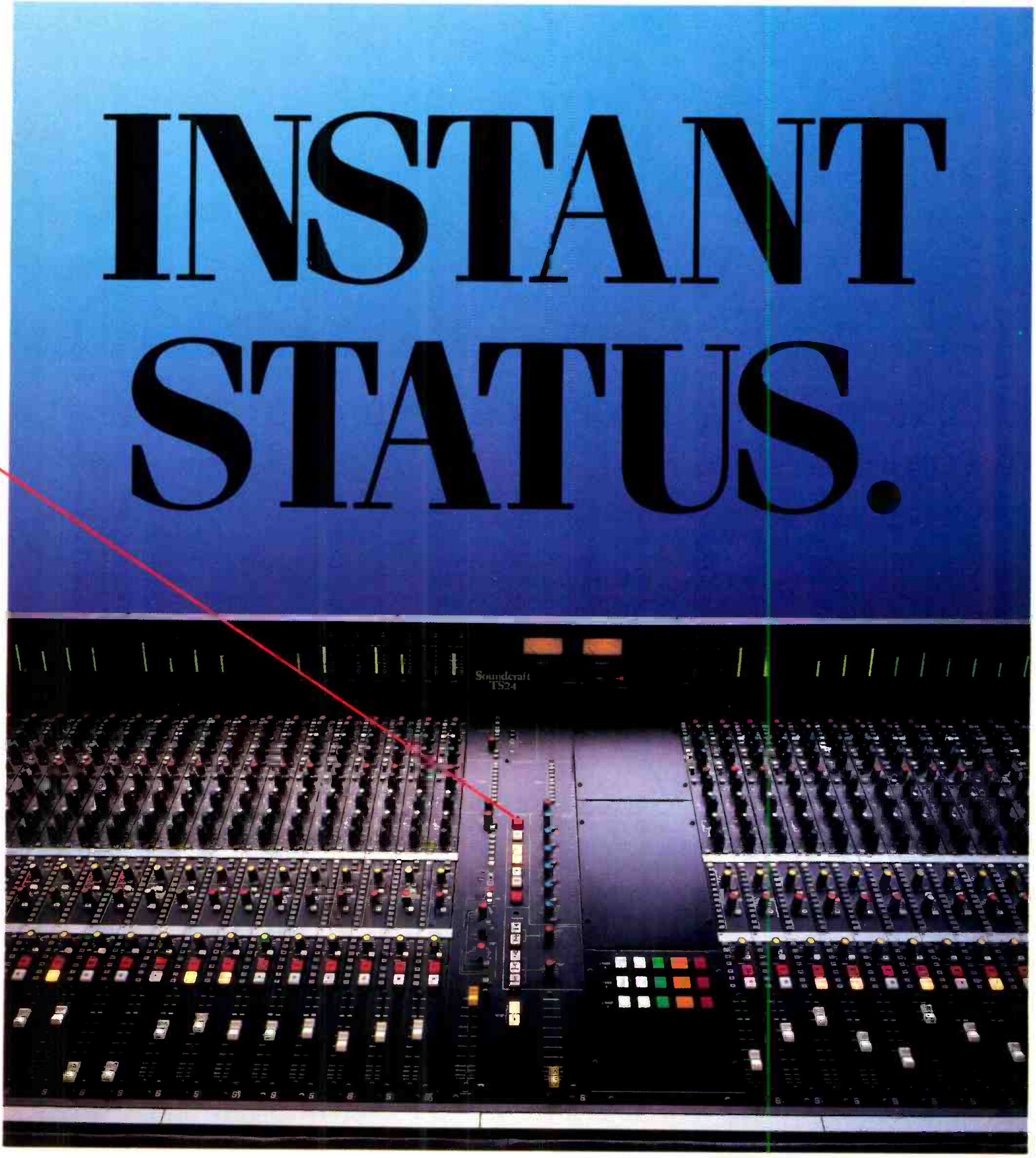
One feature of this system enables you to by-pass the Channel VCAs, thereby optimising the original recording quality.

Surprisingly enough, all this practical technology, combined with sleek good looks doesn't carry a huge price tag. So our doors are open to practically everybody.

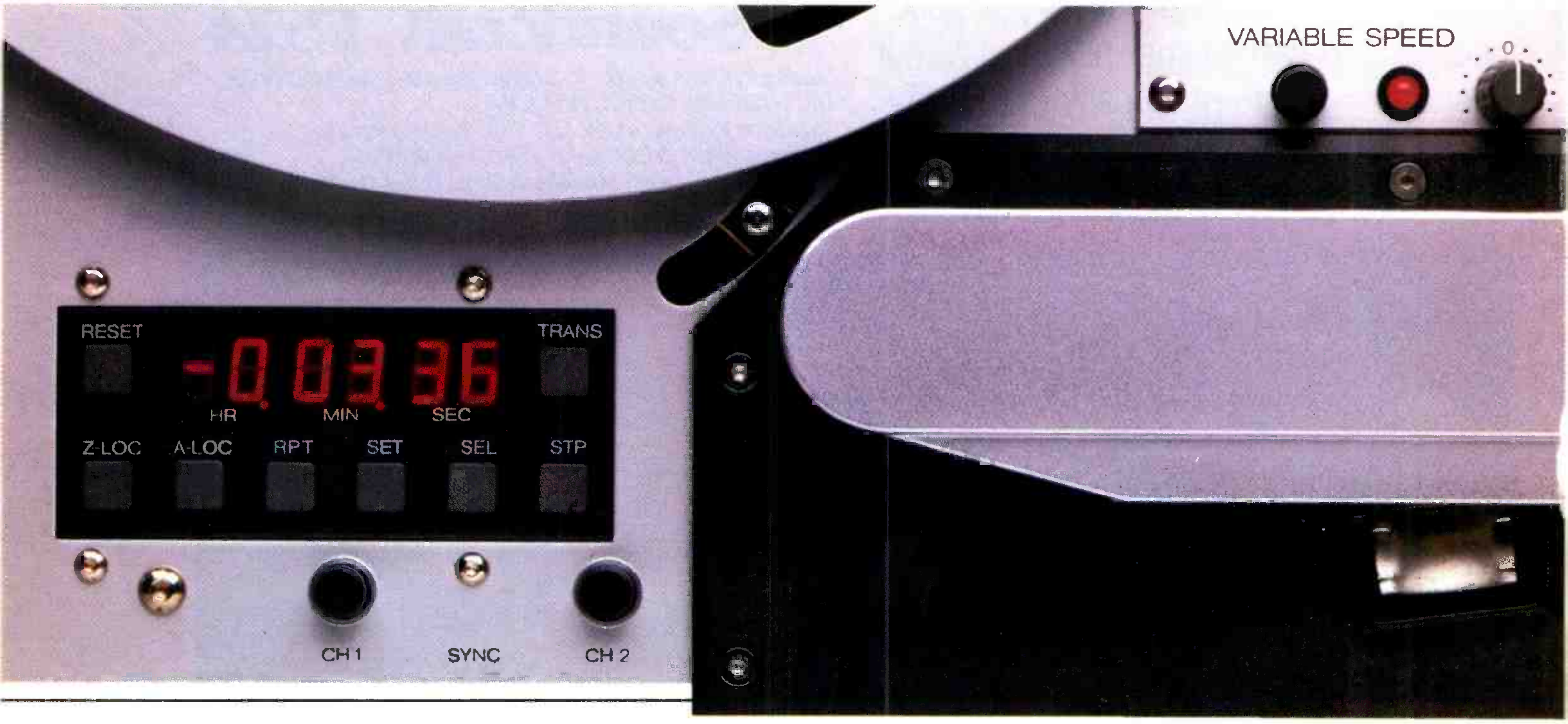
Which only leaves us with one thing to say: if you want to keep your finger on the button in the most up-to-date mixing console design available, contact us.

Soundcraft TS24

Soundcraft Electronics Ltd., 5-8 Great Sutton St. London EC1V 0BX. Tel: 01-253 6988. Telex 21198 SCRAFT G.
Soundcraft Electronics USA, 1517 20th. St, Santa Monica, California 90404. Tel: (213) 453 4591. Telex: 664923.
Soundcraft Canada Inc. 1444 Hymus Blvd., Dorval, Quebec, Canada H9P 1J6. Tel: (514) 685 1610. Telex: 05 822582.



Swiss Audio: Technical Evolution



On adding time-saving production features to a proven audio recorder design.

The updated PR99 MKII, now offering a microprocessor controlled real time counter, address locate, zero locate, auto repeat, and variable speed control, can improve your audio production efficiency. And, as before, it's built to meet strict Studer standards for long-term reliability.

Welcome to real time. The PR99 MKII's real time counter gives a plus or minus readout in hours, minutes and seconds from −9.59.59 to +29.59.59. Counter error is less than 0.5%, and the microprocessor automatically recomputes the time displayed on the LED counter when you change tape speeds.

Fast find modes. Press the address locate button and the PR99 MKII fast winds to your pre-selected address, which may be entered from the keyboard or transferred from the counter reading. Press zero locate and it fast winds to the zero counter reading. In the repeat mode, the PR99 plays from the lower memory point (zero or negative address) to the higher point, rewinds to lower point, and re-activates play mode for a continuously repeating cycle.

Pick up the tempo? When activated by a latching pushbutton, the front-panel vari-speed control adjusts the nominal tape speed across a −33% to +50% range. The adjustment potentiometer is spread in the center range for fine tuning of pitch.

Future perfect. The PR99 MKII also offers a serial data port for direct access to all microprocessor controlled functions.

Much gained, nothing lost. The new MKII version retains all features of its highly regarded predecessor, including a die-cast aluminum chassis and headblock, balanced and floating "+4" inputs and outputs, self-sync, input mode switching, and front panel microphone inputs.

European endurance. Designed and built in Switzerland and West Germany, the PR99 MKII is a product of precision manufacturing and meticulous assembly. Every part inside is made to last.

To discover more about the world's most versatile and dependable budget-priced recorder, please contact: Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

STUDER REVOX

PR99 MKII with optional carrying case and monitor panel. Roll-around console also available.

For additional information circle #9

Audible breakthrough

Here at last is a breakthrough in artificial reverberation realism. Klark-Teknik's ongoing investment in research has done more than just bring down the cost of creativity in broadcast and recording. By developing the first practical application of advanced algorithms, instigated by mathematicians in the '60's, Klark-Teknik have actually overturned the basic theories of digital reverberation to bring you "added density" reverberation.

You can clearly hear the difference!

"ADDED DENSITY"™ REVERBERATION

Natural reverberation consists of an infinite number of reflections. Conventional reverberation theory stipulated a number of reflections claimed to be adequate to simulate reverberation for the human ear.

Klark-Teknik Research first challenged this concept in theory – and have now disproved it in practice using the latest very large scale integration technology and a new generation digital signal processor (DSP) . . . its increased computing power handles information much faster than conventional hardware – without adding complexity. This gives DN780 reverberation a more natural ambience while leaving you free to concentrate on creativity.

USER FRIENDLY INTERFACING

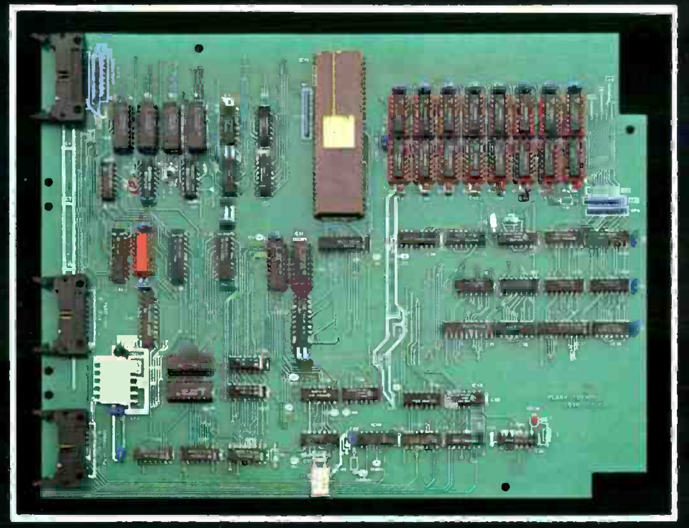
The DN780 has learned the lessons of ergonomic design. Intensive market research has led it to be truly user friendly. Intuitive combinations of clear LED displays and one-finger nudge controls make it really easy to set up new and distinctive sounds and to create unique musical environments. Any variations from the basic programmes can be stored and recalled by a simple push button routine.

IMPRESSIVE PROGRAMME LIBRARY

A wide choice of sounds can be stored in up to 89 non-volatile memory locations. Each DN780 is supplied with a free, comprehensive factory-set programme library. There are a further 50 memories available allowing you to create and store your own best settings. You can even step through a number of memory locations in user determined sequence, making the DN780 ideal for film, drama and electronic music production especially where different size room acoustic simulations with high realism are required.

COMPREHENSIVE EFFECTS PACKAGE

Better still, all reverberation programmes are complemented by a comprehensive selection of special effects, initially ranging from straight delay and multi-tap echo to a novel infinite room programme.



Advanced Digital Signal Processor.

Input headroom indication is by peak reading LED column. Illuminating the red LED indicates onset of an overload condition.

Input LEVEL control is adjustable from 6dB gain to infinite attenuation.

The photograph of the DN780 below is ACTUAL SIZE.

Green LED displays show current settings for all *seven* parameters. Push buttons under each display select that parameter for adjustment and illuminate a corresponding LED arrow. Adjustment of the chosen parameter is achieved by using the up/down buttons.

Non-volatile user-memory with battery back-up 'remembers' all parameter settings while unit is powered down. This leaves your personal sounds available at the touch of a button – even on the road.

Replaceable EPROM makes software updates easy and guarantees against product obsolescence.

Current MEMory display shows the number of the memory location currently selected.

Numeric MEMORY keyboard allows instant selection of memory location by simply entering the appropriate two-digit number. If the selected memory is empty, the display shows a broken line and then reverts to the last used memory location.

Special keys include STOre button. Pressing this and then keying a number stores the parameters currently displayed – in the new memory location. You can 'overwrite' previous settings in memory, but only intentionally – a safety interlock prevents accidental erasure of previous settings. The attempt to overwrite factory programmed settings will display a broken line and the word 'NO'.

The **SEQuence** button on the keyboard is used first to store the numbers of a series of up to 16 memories, factory or user programmed. The same key – or its duplicate on the remote control unit – will then step from one to the next in the required order, allowing rapid movement through a series of previously planned acoustical settings – ideal for film and TV production or live performance where the effects requirements vary greatly. The sequence number is displayed on the memory number display, distinguished by a vertical bar.

A MORE NATURAL SOUND

The DN780's newly developed algorithms produce a more natural sound. It uses the massive processing power of its 32 bit VLSI circuitry – the latest in microelectronic technology. This technique allows greatly increased computing speed and has added capacity to process the output of the 16 bit A/D converter without distortion. Result: highly natural reverberant sound even for notoriously difficult small acoustic environments.

ENGINEERED FOR SHOW TIME RELIABILITY

The DN780 has made another breakthrough in digital reverberation technology. By designing for better solutions, such as customised thick-film filters and low component count, it was possible to break through the roadability barrier to minimise failures while maximising performance. In short, the DN780 offers greater reliability and therefore genuine roadability.

DIAGNOSTIC CAPABILITY BUILT-IN

When the DN780 is first powered up a comprehensive self-diagnostic routine is automatically carried out, ensuring continuing accuracy in operation.

INput MUTE removes signal feed to the reverberation section, enabling the decay qualities of a chosen setting to be confirmed.

REVerb MUTE gives a rapid means of killing unwanted reverberant sounds.

PREDELAY controls the delay between the initial signal and the onset of reverberation.

REFLECTIONS
Two parameters 'pattern' and 'level' together determine the basic acoustic character of the simulated space.

PATTERN This innovative control sets the acoustic 'signature' of the environment by changing the number, spacing and density of the first reflections.

LEVEL determines the balance of early reflected energy relative to that of the reverberant sound.

REVERBERATION
'Decay', 'LF' and 'HF' give wide ranging control over reverberation time and absorption characteristics.

DECAY sets the nominal mid-band decay time.

LF varies the decay time at the low frequency end of the reverberation spectrum.

HF adjusts the high frequency decay time.

ROOM SIZE sets reverberation coefficients to simulate a room with linear dimensions in metres as shown on the display.

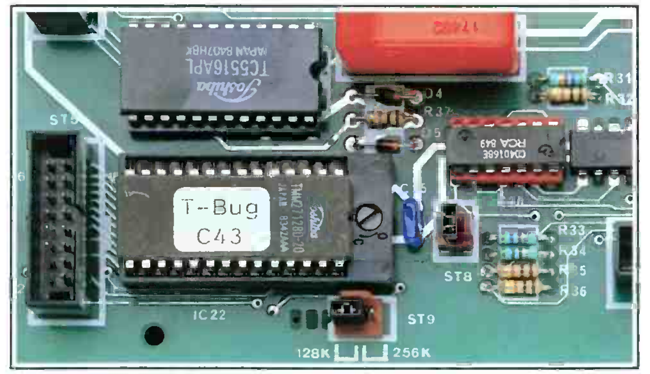
PARAMETER UP/DOWN buttons control the parameter currently selected. They change the setting in three ways – nudge; slow increment, or high speed increment.

Power on/off button is readily accessible. When first pressed the comprehensive self-diagnostic software is run, ensuring continuing accuracy in operation. After this test sequence, the unit returns to the memory location last used.

Specification DN780

Input	
Type	one, electronically balanced
Impedance balanced	20k
unbalanced	10k
Output	
Type	two, fully floating transformer balanced
Min load impedance	600 ohms
Source impedance	<50 ohms
Max level	+21dBm
Frequency response	20Hz – 12kHz +1 – 2dB
Distortion	0.03% @ 1kHz
Dynamic Range	85dB typical
Digital	
A/D & D/A converters	16 bit linear
Arithmetic Processor	32 bit
Parameters	
Predelay	0-990msec
Decay Time	0.1-99sec
Room Size	5-100 metres linear dimension
HF/LF decay	Adjustable in 14 steps (relative to 1kHz decay time)
Early reflections	PATTERN, 5 Variations LEVEL, adjustable in 10 steps (0-max)
Power requirements	
Voltage	100/120/220/240V 50/60Hz
Consumption	40VA
Weight	
Net	7.5kg
Shipping	10kg
Dimensions	
Width	482mm (19 inch)
Depth	310mm (12¼ inch)
Height	89mm (3½ inch)
Terminations	
Input	3 pin XLR
Outputs	3 pin XLR
Power	3 pin CEE
Options	Transformer balanced input PFR – Remote control

As part of a policy of continual improvement, Klark-Teknik reserve the right to alter specifications without notice.



Update EPROM in position on a 'zero insertion force' socket.

VERSATILITY WITH PLUG-IN PROGRAMMES

Ingenious programming provides unique versatility with the most naturalistic ever factory-programmed hall, room, chamber and plate settings in memories 1-20. They can be called up by the keyboard, modified with the parameter controls, and any such modified programme can be stored for later use in one of the fifty user memories. Each memory stores all parameter settings and displays these when recalled.

HALL – memories 1-5
Early reflections of low density give depth and realism augmented by slow attack and smooth decay.

PLATE – memories 6-10
High initial density and diffusion leading into smooth decay – a bright, clean attacking sound ideal for percussion and most contemporary music.

CHAMBER – memories 11-15
The uneven moderately dense early reflections produce a bright, lively sound midway between 'hall' and 'plate'.

ROOM – memories 16-20
Short high density early reflections with medium to fast attack and high diffusion produce authentic room simulation for drama, film dubbing and ambience applications.

PLENTIFUL SPECIAL EFFECTS

Sraight 0-2 sec. DELAY
This can save the cost of extra equipment by allowing the DN780 to double as a high performance delay line with easy control and really long delay time. Add regeneration to create 'repeat echo'.

Multi-tap ECHO
A high performance version of a multi-head tape echo. With a choice of 'head spacing' patterns and the benefit of digital regeneration.

Uncanny 'INFINITE ROOM'
An electronic 'zero-absorption' space in which sound is continually reflected . . . fresh input can be added to build up background in stages.

Quality ADT
A high quality double tracking facility with precise control of delay and independent control of direct signal level. Extra taps may be added to create 'choir' effects.

Versatile SOUND-ON-SOUND
A digital tape loop simulation that gives finger tip control of loop length and erasure.

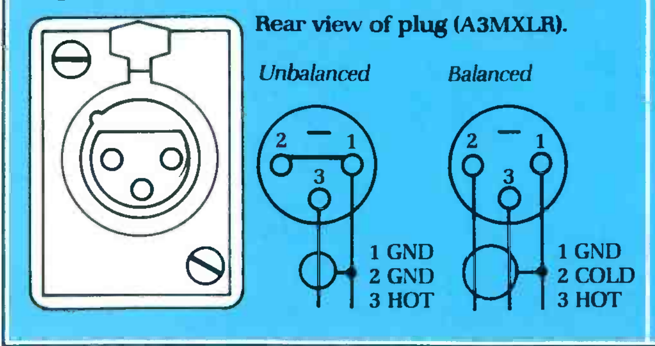
NEW PROGRAMME DEVELOPMENT

Programme development for the DN780 will be continuous over the next few years, giving even greater versatility in the future.

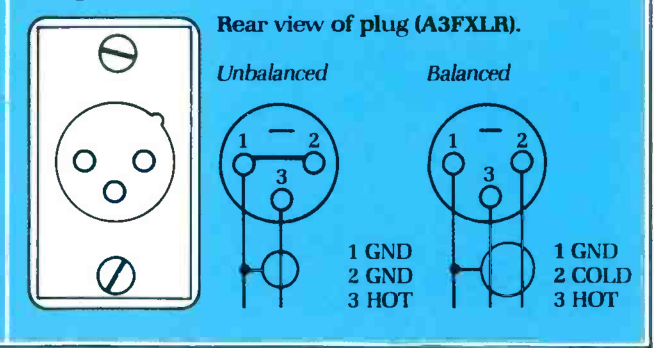
Purchasers of the DN780 are entitled to receive new programmes in the form of plug-in updates on an EPROM. Hinged circuit boards and a special 'zero insertion force' socket make it easy to fit the new programmes when they are received.

PIN CONFIGURATIONS

Input



Output



Remote Control

The compact Remote control unit can be located anywhere in the control room. Slider controls allow rapid adjustment of 'pre-delay', 'reflections level', 'HF' and 'decay time'. Each slider operates only when moved to coincide with the current setting for that parameter and this is confirmed by the appropriate **TRACK** LED illuminating. When running effects programmes all major parameters are also slider controllable.

A push button allows remote operation of the sequence control. The remote is enabled by the 'remote' push button. Pressing any parameter control on the DN780 front panel cancels remote 'on' status.

DN780 Applications

The Klark-Teknik DN780 offers many benefits for broadcast and recording studios – the most natural reverberation ever with easy control, the widest range of effects . . . and all at a cost lower than that of many less effective products.

For musicians and groups it offers the best of studio standards in reverberation plus digital effects processing.

Klark-Teknik *'reliability control'* means that every Series 700 unit – including the DN780 – is aligned and bench tested before a burn-in period and final performance test.

Series 700 Delay Lines – the big performers

The DN780 is the latest in the Series 700. The right kind of digital technology at the right kind of price for a number of very different purposes, often with features never before available.

Klark-Teknik Plc
Klark Industrial Park,
Walter Nash Road West, Kidderminster,
Worcestershire DY11 7HJ, England.
Telephone: (0562) 741515 Telex: 339821

Klark-Teknik Electronics Inc.
262a Eastern Parkway, Farmingdale,
N.Y. 11735, USA.
Telephone: (516) 249-3660

Your supplier is:

Design & Production by Nicholas J. Jones Graphics, Gloucestershire, England. 84931

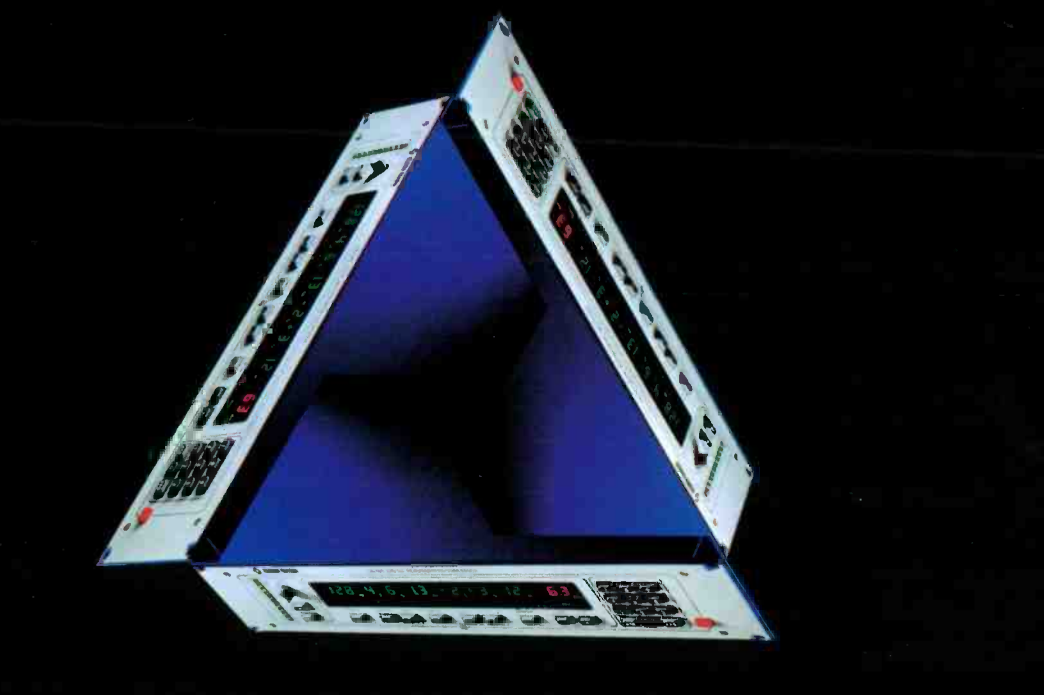
Klark-Teknik DN780

DIGITAL REVERBERATOR/PROCESSOR

The Klark-Teknik DN780 is the first of a new generation of digital reverberators combining intuitive control with a more natural "added density"™ sound.

KLARK TEKNIK

British designed, British made
Trusted throughout the World



Dedication is the soul of good design. Klark-Teknik are dedicated to making every product a classic.

Terry Clarke

TRAVELS WITH THE EDITOR

Digital Developments at the European Hamburg AES Convention

by Mel Lambert

One of the most important developments that was announced during the European AES Convention, held at the Hamburg Congress Center in early March, was news of an enhancement to the quarter-inch two-track DASH (Digital Audio Stationary Head) format. As many *R-e/p* readers may already be aware, there has been a certain amount of concern from potential users of DASH-format two-track machines that the currently proposed 7½ ips tape speed may not be appropriate for critical mastering applications — in particular, that the cueing and cut-and-splice editing capabilities might prove inadequate, and that the degree of error-detection provided at such a low writing speed could prove problematic. During a specially organized technical meeting prior to the start of the convention, representatives of Matsushita, Sony and Studer — the three companies that originated the DASH format —announced an extension to the 16-bit/48 kHz implementation that will enable 15 ips operation of two-track machines.

The new technical extension — referred to as Twin- or Double-DASH — utilizes the previously defined M-Version format of two data tracks per audio channel at 15 ips tape speed, and combines double recording with interleave delay of odd/even samples. In essence, the digital data stream is recorded *twice* along the length of the tape, with matrixing to ensure that the second data recording is made 240 blocks (around 1.6 inches) away from the first. (This is contrast to the original 7½ ips S-Version implementation, which utilizes four data tracks per audio channel, with error-correction interleaving.)

Utilizing the new 15 ips implementation, the DASH committee explained, the Twin-DASH format "will handle tape splices and heavy fingerprints without interpolation and error concealment."

While the committee explained that, in theory, upward compatibility between the two formats would be possible — with material recorded at 7½ ips capable of being replayed on a 15 ips Twin-DASH machine, but not *vice versa* — a dual-speed machine, according to Roger Lagadec of Studer, "is possible on paper, but in reality a tall order." Apparently, the record/replay equalization and filter circuitry are both different for each tape speed, not to mention different data-block formats. A better explanation of the connection between the 7½ and 15 ips implementations, Lagadec continued, would be to consider the compatability between Twin-DASH and 15 ips *four-channel* digital tapes. Obviously, a deck capable of recording a total of four audio tracks on quarter-inch tape should be capable of handling — in theory, at least — either four discrete tracks, or two pairs of duplicate data with matrixing; it remains to be seen whether hardware will be offered by the DASH manufacturers that will accommodates such a format option.

I subsequently discovered during conversations with the two companies currently offering DASH two-track machines that the Sony PCM-3102 initially will be made available in a 7½ ips version, while Studer (as mentioned below) plans to market the D820X 15 ips version first, with a 7½ ips version being made available to order.

On the hardware front, **Advanced Music Systems** unveiled the new AudioFile digital audio storage and editing system, which utilizes a Winchester disk drive to record up to an hour of 16-bit digital audio at a sampling frequency of 48 kHz. Storage time can be increased with the addition of more hard disks, the company says. The system has the potential of providing eight simultaneous outputs, each of which can be enabled independently to a quoted accuracy of one microsecond. Also featured is a built-in SMPTE timecode synchronizer, software-defined softkeys for transport control, and direct connection to the company's Timeflex unit for modifying replay timebases without pitch change.

Future systems will feature direct AES/EBU digital inputs and outputs, plus composite-video connection to 14- and 16-bit PCM processors for direct entry of digitized sound samples. AMS plans to have the microprocessor-controlled system in production by June or July.

EMT-Franz, whose products are available in the U.S. through **Gotham Audio**, unveiled the new Model 488 Digital Recording System, which utilizes a removable 5¼-inch floppy-disk cartridge to provide storage times of up to 50 seconds of mono 16-bit digitized audio at a 48 kHz sampling frequency. A total of 14 "events" or audio segments of variable length can be labelled and replayed upon demand. Intended for audio post-production, broadcast and film-sound applications, the Model 488 can be linked to additional fixed-disk units to provide extended storage capacity.

Also on show: the new Model EMT 445 digital delay unit, which features 16-bit/48 kHz operation and provides up to 10.9 seconds of stereo delay.

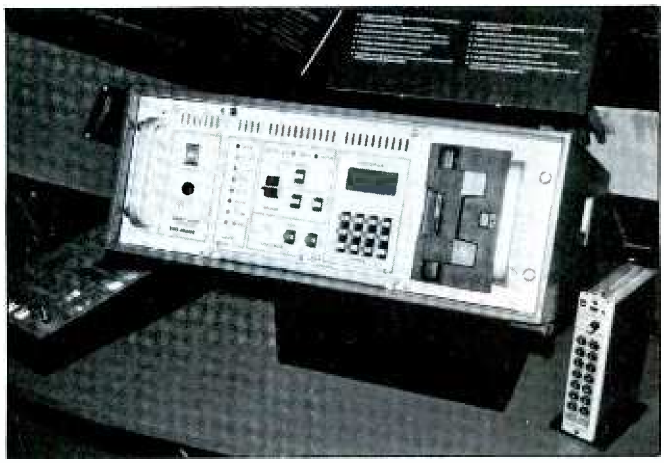
Enertec Schlumberger, the French

... *continued on page 28* —

STOP PRESS: At the NAB Convention in mid-April, Sony unveiled the new 15 ips Twin-DASH PCM-3202, based on the 7½ ips -3102; availability is scheduled for the fourth quarter of '85; price: $19,500. I understand that a dual-speed (7½/15 ips) DASH transport is also currently under consideration — *ML.*

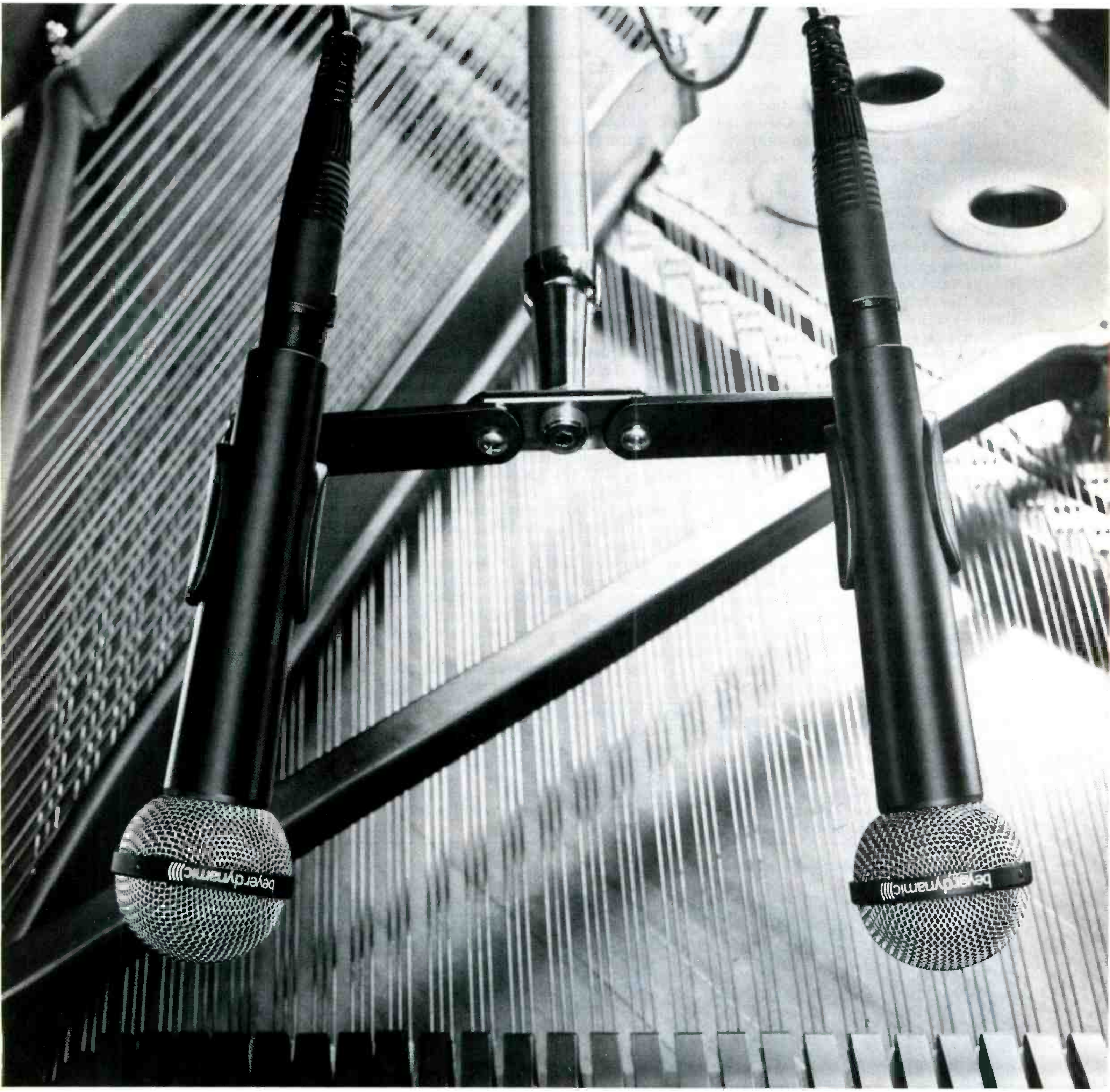
— Advanced Music Systems AudioFile System —

— EMT-Franz Model 488 Digital Storage System —



KLARK TEKNIK SIX-PAGE INSERT PRECEDES ON PAGES 19 thru 24

BEYER RIBBON MICROPHONES AND



THE DYNAMIC DECISION

— Entertec Schlumberger DAS Digital Console —

— RTW PCM-Set2 and PCM-Set 3 Interfaces —

— continued from page 25 . . .
manufacturer of consoles and tape machines, was showing an interesting prototype of its DAS all-digital mixing console, which is expected to be marketed within two years. The 16-input/four-group version on display in Hamburg featured a central assignable control section that included a three-band, fully parametric equalizer, four auxiliary sends, group routing, plus mike/line input selection. Direct digital inputs and outputs are standard 16-bit at a 48 kHz sampling frequency.

Production versions, the company says, will offer up to 64 analog/digital input channels, 32 main groups, 16 auxiliary/effects sends, and flexible sub-mastering capability.

HHB Hire and Sales announced enhancements to its CLUE (Computer Logging Unit and Editor) microprocessor-based system for use with the Sony PCM-701ES digital processor. The system now includes the ability to edit between PAL and NTSC composite-video formats with a modified 701; a SMPTE/EBU timecode reader card; switch-selectable U-Matic/Betamax VCR operation; and a printer output for producing hardcopy of edit decision lists and timing data. It was also announced that the CLUE System is now being marketed in the U.S. through **Audio Intervisual Design**, Los Angeles.

RTW, whose products are available in the U.S. through **Auditronics, Inc.**, was demonstrating the new PCM-Set 2, which comprises a modified Sony PCM-701 digital processor and an interface unit for digital copying and conversion from EIAJ 16-bit to 1610 format. An enhanced model, PCM-Set 3, provides time compensation between left and right audio channels, and offers transfer from 1610 to 701/F1 formats. U.S. pro-user prices are $2,225 and $2,675 respectively; there are also small additional charges, Auditronics points out, for internal modifications necessary to the PCM-701 processor.

Studer/Revox America, Inc. announced that the D820X digital two-track based on the new 15 ips Twin-DASH format — details of which are provided above — will be available from mid-June. The initial production run of 25 machines has already been pre-sold, however, and the D820X is currently on back order until January 1986. The digital two-track is expected to cost between $20,000 and $25,000, depending on system options.

Also to be seen: the new Studer 969 console, an updated version of the 169/269; a TLS4000 timecode synchronizer interface for the A80VU Series multitracks; and a production version of the A820 analog two-track, deliveries of which were scheduled to begin in mid-April.

Denon/Nippon Columbia Co., Ltd. was showing a prototype Model DN-039R PCM processor that enables two or four digital audio channels to be recorded in NTSC composite-video format on a standard U-Matic videocassette. The encoding format is one developed by Denon, and offers 16-bit resolution at a sampling frequency of either 44.1 or 48 kHz. While the four-channel mode of operation was deve-

. . . continued on page 33 —

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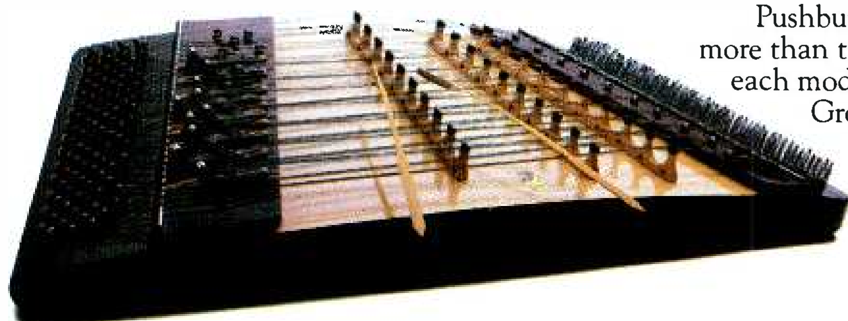

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Other system elements include events control, programmable equalisation, and a variety of mainframe and metering options to suit many different requirements and budgets.

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— continued from page 28
loped primarily for classical recording, the two-channel format utilizes double recording of interleaved data for improved reliability of the encoded bit-stream. Details were also being provided on the new DN-050MD all-digital console that has been in use for CD mastering at Denon's facility in Japan since the summer of 1984. Configured as a four-input/stereo-out console, the unit features analog plus direct AES/EBU digital, Sony PCM-1610 or Denon-format composite-video inputs and outputs. Also provided for processing signals in the digital domain are programmable EQ functions, two channels of compression, limiting and expansion, DC offset cancellation, pre- and de-emphasis correction, automated fade-in and -out, phase reversal, and full metering. The system is expected to be available in the U.S. by the end of the year.

Other items of interest at the Hamburg AES Exhibition:

• **Audio Kinetics** announced that its range of Q.Lock timecode synchronizers has been redesigned to accept a total of four audio and video machine interfaces. The new 4.10 retains the same control unit as the 3.10 for simultaneous operation with three audio/video transports, but the system configuration can now be selected by the user at the computer frame. In addition, space has been provided in the enlarged frame for an optional VITC reader, plus additional accessory cards such as the 16-event relay option. Present 3.10 users will be able to upgrade to the 4.10 format with a new frame and interface. Also being demonstrated: the new Eclipse Editor that functions as an intelligent controller to offer full four-machine control with the 4.10 synchronization computer. Features include a 20-line miniature VDU screen for displaying machine status, timecodes, and set-up menus; the ability to establish two independent groups of transport configurations, plus individual control if required; 12 user-definable softkeys to set up control sequences; and 100 loop memories for EFX and ADR applications.

• **Klark-Teknik Electronics** unveiled the new Klark Acoustic System 2.1 powered loudspeakers, which are said to combine high SPL in a compact, fully integrated design. A 27-liter, two-way bass reflex configuration is featured, with a built-in MOSFET amplifier and active crossover. Designed for close-field monitoring and portable applications, speaker pairs are quoted to be matched to within 0.5 dB, and offer a 50 Hz to 18.5 kHz response within 3 dB. Price will be in the region of $1,650 per pair, including telescopic stands for vertical and horizontal operation.

• **Otari Corporation** reported that is currently delivering the MX-70 eight- and 16-track on one-inch machine. Also on display: a final production prototype of the new MTR-20 mastering deck, available in quarter- and half-inch/two- and four-track versions (the former with optional center-track timecode), with full microprocesor control of deck and electronics functions.

• **Quantec**, whose products are available in the U.S. through **Europa Technology**, was demonstrating the new QRS/L Room Simulator, which comprises a mono-input/two-output version of the well-known two-in/four-out QRS model. Identical operational and memory features have been retained on the new "mini-version."

• **UREI** unveiled the new Model 809 monitor loudspeaker, described as a smaller version of the well-known 813 unit and intended for close-field monitoring and use in smaller control rooms. The Time-Aligned unit features a new custom-designed 12-inch coaxial driver with a 1.8 kHz crossover frequency.

• **Ursa Major** was demonstrating the new MSP-16 digital Multitap Stereo Processor, which is capable of ambience and stereo synthesis, plus various special effects. Front-panel controls provide selection of any of eight operational modes; additional plug-in PROMs will extend the number of modes to 16. Pro-user price is $2,000. Also on display: the Stargate 626 digital reverb and special effects unit, which is an enhanced version of the previous Stargate 323 featuring a total of 16 mode settings and an increased delay capability of two seconds; price is $2,500. ■■■

EXPOSING AUDIO MYTHOLOGY

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

by John H. Roberts

This being the April column, I considered a fictional analysis of OFHC (Oxygen-Free High-Conductivity) wire. I could have argued that there was no oxygen in vacuum tubes, so why should there be any in wire? (But, after a few equally bad jokes, I decided it just wasn't that funny.)

My interest in OFHC wire was renewed by a recent product introduction from a major manufacturer of Compact Disc players. This latest-generation CD player, available at a very reasonable $450 price tag, is reported to use OFHC wire in its audio path.

I appreciate that manipulating the crystalline structure of metals in wire can reduce series resistance, but doubt that even a "super conductor" (zero resistance) would make an audible difference in such a short run. I wonder how long before we see Litz-wire line cords? (Note: There are certain applications where resistance weight versus cost is an important factor — it's just that they don't occur in earthbound hi-fi gear.)

As usual, I invite responsible discourse on this subject. Now on to the non-fiction part of my column.

Digital Tape Life

On the subject of just how rugged is digital data recorded on magnetic media (October 1984 issue of *R-e/p*), the best (only) information I've come up with so far is a report of a National Bureau of Standards Study. The report stated that tapes stored at optimum temperature and humidity may last as long as 20 years, but that excessive heat or incorrect humidity could reduce that time to as little as five years.

Just like any other magnetic recording, however, a stray magnetic field ("I thought I turned the head demagnetizer off!") can reduce that to five minutes. So,I guess if you have a really valuable recording the best way to preserve it for posterity is to cut a CD.

Discrete Versus IC Designs

It has been argued by some designers that discrete-circuit designs are audibly superior to integrated-circuit designs. This contention is worth reviewing because all-discrete designs are swiftly going the way of the vacuum tube. But are we compromising sound quality for economic considerations?

I have been careful to couch this question in terms of "sound quality" and "audible" effect. A carefully executed discrete design will usually exhibit a wider dynamic range than the typical version. Discrete designs can swing to arbitrary high output levels, and offer several dB less noise than the best integrated op-amp. On paper discrete wins but, in the real world, the advantages are somewhat diminished since audio signals do not approach the dynamic range of even the better "jelly bean" (TLO74, NE5534, etc.) op-amps.

While it is clear that integrated circuits can handle the lion's share of audio applications, several functions are likely to remain discrete for at least the near future.

First: At present there are no integrated circuits that can effectively deal with the 50- to 150-ohm source impedance of professional microphones. I have seen IC pre-amps with less than 1 nV/rt. Hz (that's more than 13 dB quieter than a 5534!), but with unacceptable linearity. Discrete transformerless design and op-amps with suitable step-up transformers can beat that by an immeasureable margin.

Second: Another function notably populated by discrete components are output drivers. While most line-level processing gear — including digital recorders —will never need to see more level than can be delivered by a 5534-type op-amp, 50V peak-to-peak output swings are often sold in the name of headroom. With the possible exception of terminated-impedance systems (see my December 1984 column) a signal swing of more than 30 Vp-p is wasted. The maximum signal that the chain can pass will be limited by some internal stage running off ±18 volt power suplies.

Third: A function commonly performed by discrete devices is the summing bus in large console structures. The casual analysis of a large summing bus might point to op-amp noise as a problem, since a 48-input bus would boost the summing amp's input noise by approximately 34 dB. However, the perfect (noiseless) summation of 48 equally quiet channels will still result in a 17-dB noise increase. Since there will be usually one or more channels that are 20 dB noisier than even the popular integrated op-amps, noise isn't the real problem. Giving up 34 dB of loop gain margin can result in less than ideal phase and linearity performance. (See my April 1984 column for more details on the subject.) On a side note, at 20 kHz a TLO74 haas about 50 dB of open-loop gain; a NE5534 about 70 dB.

The Verdict

For now there are a few audio functions that are better served by discrete devices, but the majority can be, and are, easily handled by the "jelly bean" op-amps.

The difference between discrete- and integrated-circuit performance is less a function of SOTA engineering than it is simple economics. As with the marketing of any product, the more specialized a device becomes, the less people can use it; and the less people to buy it, the more you must charge. The more you charge, even less will buy it, and so on down the slippery slope. It should come as no suprise that most audio integrated circuits are designed for consumer applications. For example, the lowest noise transistor that I could find a few years back when designing a microphone pre-amp was intended for moving-coil phono pre-amps (10- to 20-ohm source impedance).

Curiously, the input differential pair of a popular discrete op-amp is actually an integrated circuit. The LM394 contains something on the order of a hundred small-signal transistors connected in parallel to form two extremely linear compound transistors. While the LM394 was designed for high-accuracy, computational "logging" circuits, a favorable side effect was low noise. The LM394 became popular in the Seventies for low-noise design because the transistors when offered as low noise weren't anywhere near the LM394's 1 nV, and the only other competition, power transistors — low noise as a consequence of low base-emitter resistance — suffered from batch-to-batch variations and process related noise.

Since then we've seen discrete FETs pushing 0.7 nV/rt. Hz, with discrete bipolars in the 0.4 nV ballpark. IC op-amps are getting down to a respectable 2.5 to 3 nV. I see no theoretical reason why an op-amp couldn't be as quiet as discrete devices. However, it would require a lot of silicon, and would be very expensive. I don't see much pressure to advance the present SOTA quickly, since the sales volume to justify it just isn't there.

On the subject of getting more power out of ICs, appliance-controller chips are already handling 100V signals. Likewise, "super-computer" research has been able to dissipate hundreds of watts in an IC package. Theoretically, it would be possible to put your PA on a chip but, economically, it won't happen — at least not for a good while.

A Parting Shot

Before I close I would like to take another swipe at the hyperbole-prone tweaks on the consumer side. Another seller of CD players very proudly proclaims the use of discrete components within its unit. I wonder if anyone has bothered to count how many discrete components it would take to make a functional CD player? Or how many trims it would take to make it work as good as the present highly integrated versions? ■■■

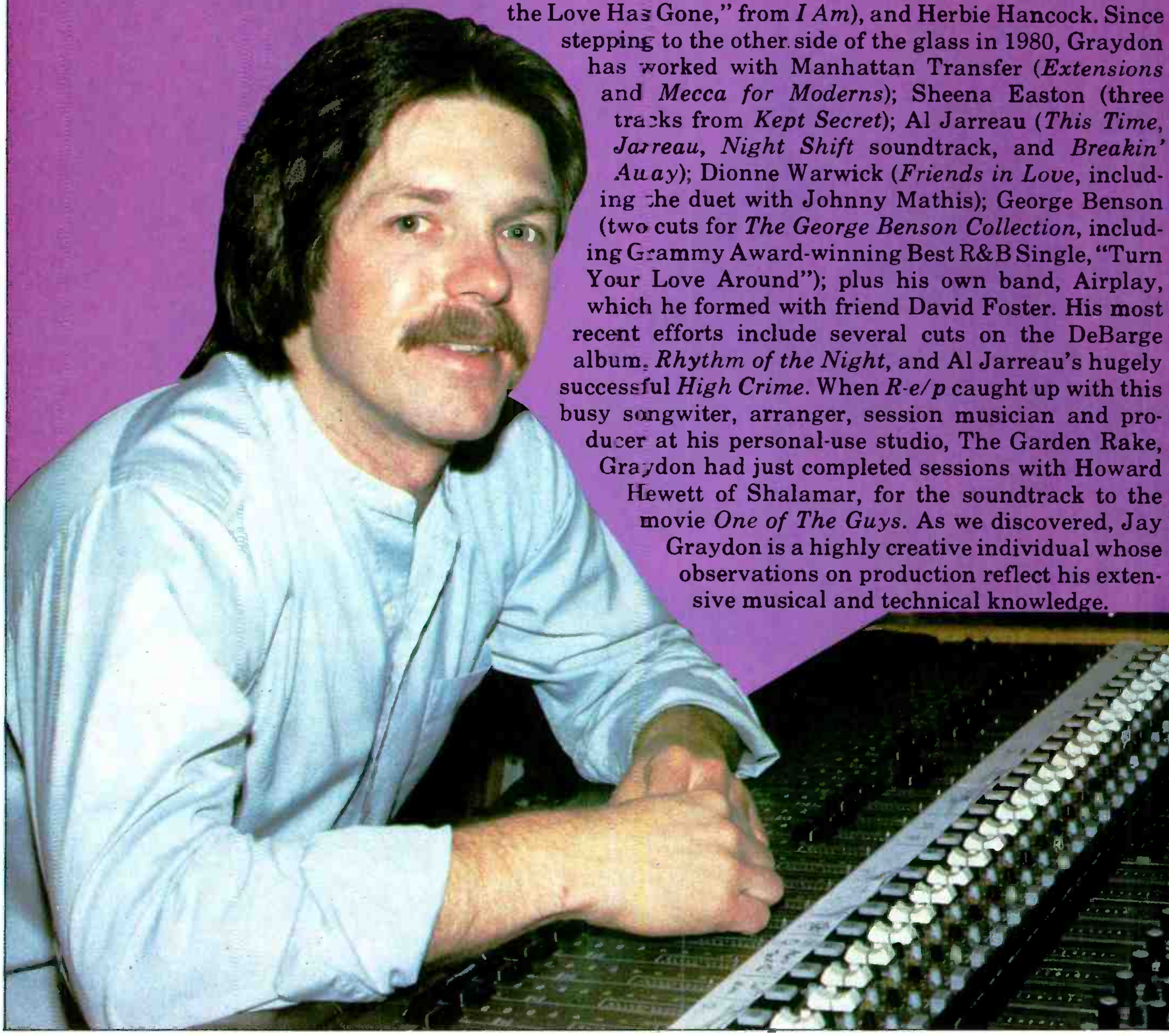

AMPEX
GRAND MASTER 456

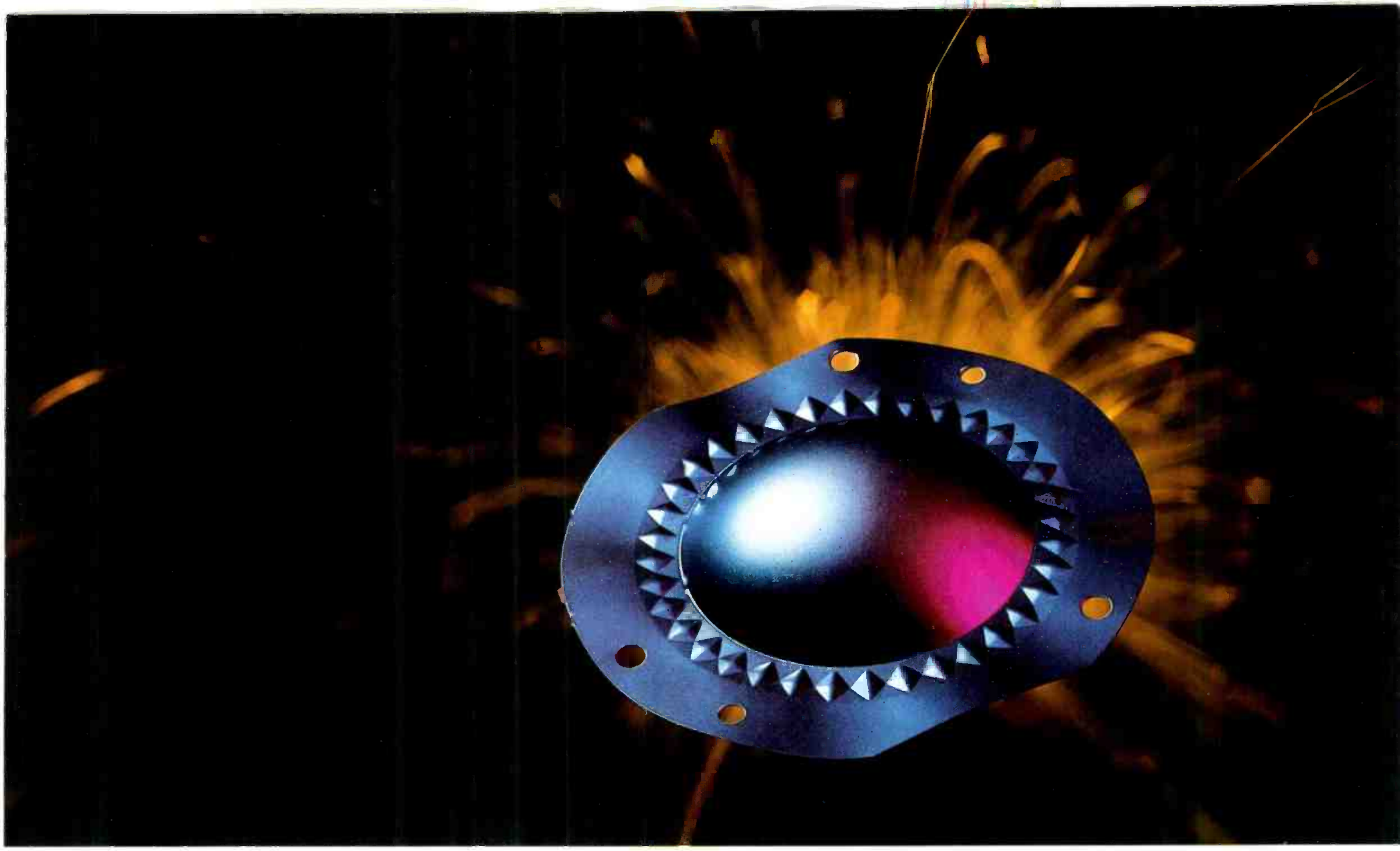
Musician/Producer/Studio Owner

Interviewed by Mel Lambert and Ralph Jones

JAY GRAYDON

In his 15 years of involvement with practically every facet of the the music business, Jay Graydon has displayed an impressive range of talents. The son of Forties-era entertainer Joe Graydon, at high school Jay played trumpet, flute, saxophone, bass and organ, and then studied music theory at Valley Junior College. He began his musical career as a guitarist, performing during the early Seventies with Gary Puckett and the Union Gap, and later with Spiral Staircase. Subsequently, he earned an enviable reputation as a first-call session musician and songwriter — he is quoted as working on no less than 800 sessions during 1977, and broke Larry Carlton's previous record by playing a total of 28 sessions in six days. His session credits include dates for just about anyone who's anyone in the business, ranging from Shaun Cassidy to Dolly Parton to Barbra Streisand to Steely Dan. His extensive catalog of songs has been recorded by such artists as Paul Anka, Dionne Warwick, Air Supply ("I Can Wait Forever," from the *Ghostbusters* soundtrack, co-authored by David Foster and Graham Russel), Earth, Wind and Fire (co-writing credits on "After the Love Has Gone," from *I Am*), and Herbie Hancock. Since stepping to the other side of the glass in 1980, Graydon has worked with Manhattan Transfer (*Extensions* and *Mecca for Moderns*); Sheena Easton (three tracks from *Kept Secret*); Al Jarreau (*This Time*, *Jarreau*, *Night Shift* soundtrack, and *Breakin' Away*); Dionne Warwick (*Friends in Love*, including the duet with Johnny Mathis); George Benson (two cuts for *The George Benson Collection*, including Grammy Award-winning Best R&B Single, "Turn Your Love Around"); plus his own band, Airplay, which he formed with friend David Foster. His most recent efforts include several cuts on the DeBarge album, *Rhythm of the Night*, and Al Jarreau's hugely successful *High Crime*. When *R-e/p* caught up with this busy songwiter, arranger, session musician and producer at his personal-use studio, The Garden Rake, Graydon had just completed sessions with Howard Hewett of Shalamar, for the soundtrack to the movie *One of The Guys*. As we discovered, Jay Graydon is a highly creative individual whose observations on production reflect his extensive musical and technical knowledge.





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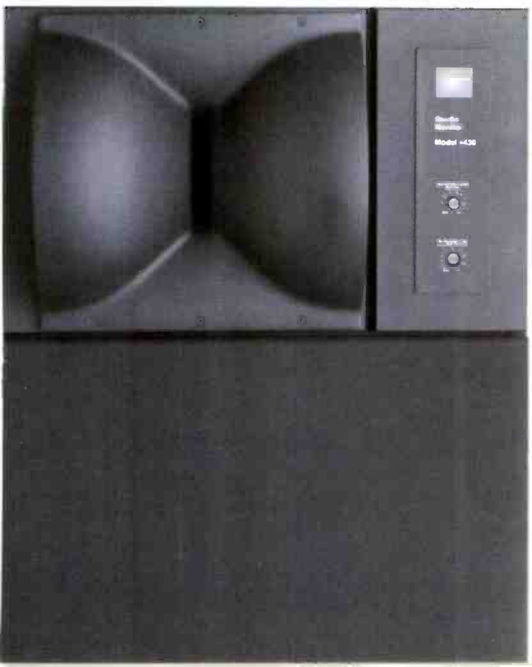

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JAY GRAYDON

R-e/p (Ralph Jones): How did you make the transition from first-call session player to producer?

Jay Graydon: I began producing in 1978, around the same time as David Foster and Michael Omartian; that's about the time that the "musician-producer" started to be much more dominant than the non-musician producer. Steve Kipner, the songwriter, was the first to approach me. He was looking for a musician to produce him, although he didn't really want a producer *per se*. He said, "You're an arranger, and a producer is basically an arranger with song sense." I'd been waiting to do something like this because, by that time, I had my own studio, and I was always into engineering. The first year, I did well. I had the Steve Kipner album, and then I did a Mark Jordan album on Warner's.

R-e/p (RJ): While you obviously have a "Jay Graydon" playing style, do you think that you have a "production sound?"

JG: No. When you're successful with somebody, then every record company figures *that's* the kind of act that you should do. But I want to do rock and roll bands . . . I want to do different things. I *don't* want to just get pegged with slick, pop records.

R-e/p (RJ): As a producer, do you find yourself analyzing other people's work?

JG: All the time, but I try to keep it to myself. [Laughter] I learn from it. The first thing you've got to do in this business is get rid of your ego in dealing with other people — they do things *their* way. If they've got a hit, and you're putting it down, why are you doing that? I want to look for the good things they did, and grow off of that.

R-e/p (Mel Lambert): Of the various producers you have worked with in the past — and maybe those you haven't — whose work do you respect most?

JG: First of all, my buddy David Foster, and Michael Omartian; I've learned a *lot* from these guys. I also admire Trevor Horn. Richard Perry has a song sense that nobody else has. Steve Berry also has a song sense that I really like.

R-e/p (RJ): You have a considerable reputation as a session player. On a tracking date, do you tend to be very directive with musicians?

JG: Yes. They get bugged with me, because I usually put them through the mill. For me, there are three things that are important. First of all, the *feel* has to be there. Second, it's got to be *tight* — that takes longer to get. And the third is that *pitch* has to be good. The feel's the most important, but they all matter to me. If I were doing a rock and roll band, where pitch and tightness wasn't a factor, all I would care about would be the *attitude*.

People put Toto down for being "slick." Well, what does "slick" mean? Is there something wrong with having a song that sounds tight? Their music sounds good in tune and tight, and it feels good!

R-e/p (ML): But wouldn't you concede that a production can be too "glossy," and lose all its life? Maybe not with Toto, but the adjective has been used with other bands who make music that sounds slick and lifeless.

JG: That's true but, like I say, the most important thing is the attitude and the feel. If the track is feeling too glossy, and it's in tune and sounds tight, then I've got the wrong band! I'll cut it again. I've cut tracks a lot of times with different studio musicians to get the right feel. In the case of bands, I can't think of an instrument that hasn't been replaced by a studio musician — it's done anonymously. But if I were recording a rock and roll band, why would I replace a musician if that band was signed because of their attitude and feel? Take any good Metal band: they don't bring studio guys in. They're *bands*.

R-e/p (RJ): I'm particularly interested in the concept of capturing the "feel" of a track. When you first begin the session, how do you go about establishing that possibly nebulous quality? How many instruments do you track on the basics?

JG: I usually start with a trio or a quartet. I spend a lot of time up front writing out the tune. I'm typically the arranger — not always, but typically. I usually have a piano player on the date, such as David Foster or Omartian or Robbie Buchanan, who are good arrangers as well. Good piano players make the best arrangers, because they can play it all at the same time.

A lot of times when I write these tunes, I don't really have specifically what I want the drummer to play. I want the drummer to feel what's comfortable for him first, and then from what he's playing I'll give him direction. We just start fooling around with feel. I'll write a bass line guide: I'll write out the chart verbatim from what I'm feeling at the moment. But I'll always say, "Look, this is where we're starting from. If you don't think it lays right for you, or you've got an idea, let's work on it."

Getting the "feel" for the tune is just hit and miss; you can't say how you're going to approach it. If I wrote the tune, I know what I want 90% of the time. But as soon as the musicians get in there, they're going to come up with ideas. You just experiment.

If the track feels good, and it's really organized, you do fewer overdubs: it sounds more like a band right off the bat. If I had a track that was not quite as organized as I want, I'd probably bury some instruments: yank things back and have some synth or guitar overdubs. But then it starts getting harder for me. Which is why I like real good basic tracks. We might work eight hours on one track; sometimes we get it in three hours. Sometimes if the track isn't working for me I'll erase everything, and start from the drums up.

R-e/p (ML): But you prefer to have as much internal cohesion as possible on the basic tracks, rather than put the song together piecemeal?

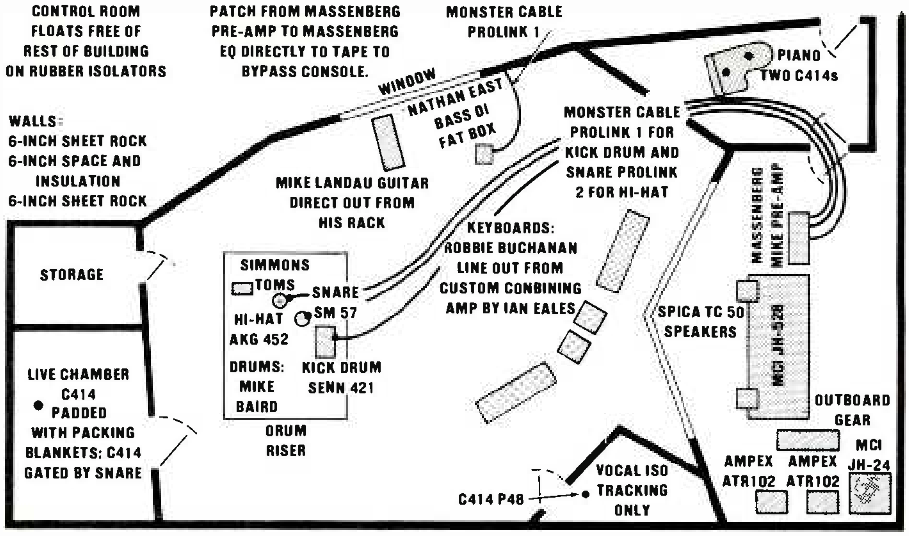
JG: Always. If I put it together piecemeal, I concentrate on each part too much. That's the problem with drum machines — you concentrate on each individual sound, and lose sight of the overall feel.

R-e/p (ML): Do you try and use the room to give you that cohesion — to create a complete environment?

JG: Right. We have a live chamber in the studio [Garden Rake, Graydon's personal-use facility] and we've got the drum stand near enough to the room that it sounds good. There's not too much of a time lag, and if there is a lag I'll flip the tape upside down, delay it and have it come early, then put it on another track. In that room, the snare sound is great. *. . . continued overleaf —*

"Material is the *whole* game: if you don't have good songs, you've got nothing! There are very few times that artists make it on momentum. Although the momentum of an artist can carry bad tunes sometimes, it doesn't happen often. And a bad tune will *never* break a new artist; you've got to have hits."

JAY GRAYDON



ROOM AND MIKE LAYOUT FOR "JUST ONE OF THE GUYS," BY SHALAMAR. STUDIO: GARDEN RAKE. PRODUCER: JAY GRAYDON. ENGINEER: IAN EALES.

R-e/p (RJ): You've been working for some time now with engineer Ian Eales, who was introduced to you by David Foster. Can you describe your working relationship with Ian?

JG: Ian is the "tweaker." We track basics together — we come to an agreement, and we both "bend" in the sort of things we're looking for. Then, I basically do the overdubs. If we're tracking synths or horns, and there's a lot of confusion, Ian might be there.

R-e/p (ML): Once the tracking is completed, do you mix the tracks yourself, or do you work with Ian? Let's use Al Jarreau's High Crime *album as an example.*

JG: Typically, Ian mixes with Michael Verdick at Channel Recording [a studio based in nearby Burbank, CA]. They bring copies of the mix to me here, and I give them direction. Ian and Verdick work well together, and I'd rather not mix the tracks. By that stage in the record, I'd like to be more objective. I also want to let things go more on emotional levels, rather than worrying about getting everything placed well.

If they don't mix the track, then I will. Ian might be involved but, at that point, when I'm so tuned into my trip, I get protective of it. And it's actually a bad thing to have happen — which is why I'd rather have them do it at Channel.

R-e/p (RJ): How do you describe the kind of mix that you want? What sorts of terms do you use?

JG: I give them the concept of the song, starting with the floor of the tune. I'll say, "The floor in this tune is going to be pretty even to the track. It's not a dance tune; I don't want 80 million dB of foot and bass, with no surrounding instruments." I'll also tell them what they'll have to split off because of EQ moves and pan moves.

I pretty much tell them the pans; we have a method for writing down pans so that we can duplicate a mix in the future. I have seven basic pan slots: center, -3, -5, -7, -10, -15 and full, sometimes -20. I run a 1,000-cycle tone into the module, and set the pan so that one side is zero on the VU meter and the other is -5, for example. If we have a single instrument that's not going to be stereo'd off, I'll just put it where it feels right and check it when I'm done; I'll connect the oscillator and see where it ended up. Sometimes it's hard to write down exactly where it is, but it really doesn't matter to be that exact.

R-e/p (ML): But that technique would only work on a board you know well, and where the metering is accurate from session to session. If you go to another board, the pan and metering may not be the same.

JG: That's true. On the [modified Trident TSM] console at Channel, -5 sounds different from -5 on my MCI console here. But they know how to deal with it; Ian knows both rooms, so he makes up for it.

So, we work on the floor of the basic tracks first, until we get it right. I'll say, "The foot's too wide now," or "You've got it a little too thick. Instead of hitting it at 100 hit it at 60, because I don't want it to speak in the bass area." "More point, less point," that kind of thing. "Crunch the bass a little harder; it's too uneven." "Make it smaller, make it more defined." We talk about the floor in those terms.

After the floor's straight, we talk about the surrounding instruments. Then I really give them the concept of the mix. I say, "Okay, at the chorus, I want the whole mix to *leap*. Somebody put their finger on the master fader, and give me 3 dB at the chorus on the whole band."

Finally, we start fine-tuning everything. I start telling them the vocal lines that are too buried or too loud. We really

RAPPORT IN THE STUDIO:
A Conversation with session engineer Ian Eales

Recording engineer Ian Eales has been Jay Graydon's right-hand man for the past four years, serving both as his engineer and as manager of Garden Rake, the musician-producer's personal-use studio. Eales represents an example of an extremely rare breed: he is a practical audiophile who also gains his living as a working audio professional. Eales' independent album credits include *Patti Austin, All I Need* with Jack Wagner, *Lifetime Guarantee* with Michael Johnson, and *Best of Me* with David Foster.

Educated at the University of Victoria, British Columbia, Eales majored in physics, but dropped out to become a professional photographer. How did he make the transition to his current metier, we asked? "I'd always had an interest in electronics, recording, and sound," he replies, "and my brother-in-law is [producer] David Foster. I had always given him a hard time about the sound of his records, and one day he said, 'Well, if you're so smart, why don't you come to L.A. and do something about it?'

"I'd never considered it before but, the more I thought about it, the more it made sense. So I sold my business, moved to Los Angeles, and he hooked me up with Jay, who was building Garden Rake at that time."

Given a significant leg-up by Foster's auspicious recommendation, Eales says he learned the craft of recording on the job. "The first project we did was George Benson's 'Turn Your Love Around,' which we recorded at Garden Rake. Jay had produced and engineered on his own before; I basically came on as his assistant. The more I learned, the more I did."

Complimentary Skills in the Control Room

"When I hooked up with Jay, much of what he wanted me to do was improve the sound of his records — make them bigger- and better-sounding. The first thing I did was to change his reference speakers. The ones he was using were very bass-heavy, with a 'screechy' treble and a deficient midrange. We changed to Yamaha NS-1000s, which we're about to change again; I also changed his power amp.

"When we began working together I was second engineer, and audiophile consultant in terms of the sound. Ours is a give-and-take working relationship: I'll want to hear something for sonic impact, and he won't because of a musical chordal relationship. We hear differently: the records I make by myself are *not* like his sound. He calls his sound 'definition;' I call it small. He calls mine 'bottomy;' I call it big. It's just a different perspective! [Wry laughter]

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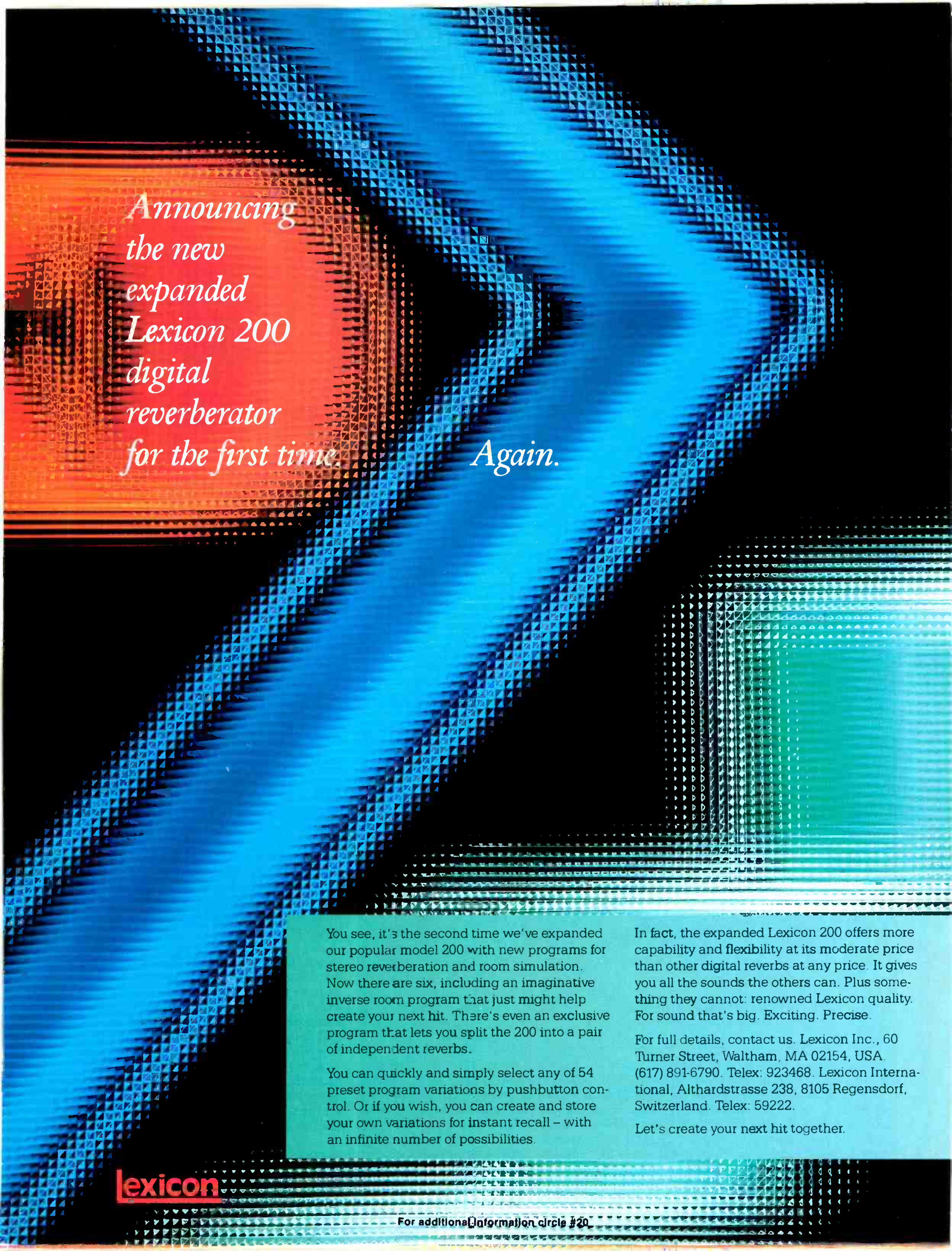
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RAPPORT IN THE STUDIO — continued . . .

"Often, my input has been only on sonics; at other times, I've done all the engineering. I've also built custom equipment for him. He likes to combine vocals, and I built a couple of devices so he can do that more easily. One is a switching network that eliminates the need to continually change patch cords. The other is a device that allows you to select and cross-fade between tracks, so you don't have to punch: breaths and other vocal nuances become much more natural if you cross-fade, and you don't have to ping-pong between tracks.

"When we do a tracking date, Jay always sits at the board, and I reach around and help. Jay is a *monster* musician, and knows what he wants — he has a definite concept of how he places things in the mix. Working with Jay, you make moves of a half to a quarter of a dB."

What sonic characteristics does he try and capture in basic tracks? "I like the drums as big as I can get them," he offers. "If the bass is live, rather than synthesized, I want it to have a solid fundamental, and be able to 'speak' while at the same time holding up the track.

"It used to be that everything went down dry. Now, we use so many effects that there just aren't enough sends and inputs on the console to handle it all. So, more effects are used on the tracks. Sometimes, I'll use the console EQ section without adding EQ: just switch the module in. It adds a distortion that's consonant with what's going through it, and makes the sound 'bigger-badder-better-meaner'!"

"Things that have to be perfect, for me, are limiters and equalizers: they should do *only* what you want them to do. For example, the [UREI] LA-2A is my favorite limiter. The opposite of that limiter is the 1176: I refer to it as a 'low filter.' It softens the bottom-end so much, but sometimes you *want* that; on guitar, for example, that's *the* sound. The [Eventide] Harmonizer is the *perfect* musical-instrument processor — it does what we all want.

"The console should simply be *totally* transparent; I always want the outboard gear to foul up the sound, *not* the console. Likewise, I want the mike lines from the studio to the control room, as well as the tape machines, to introduce minimum coloration."

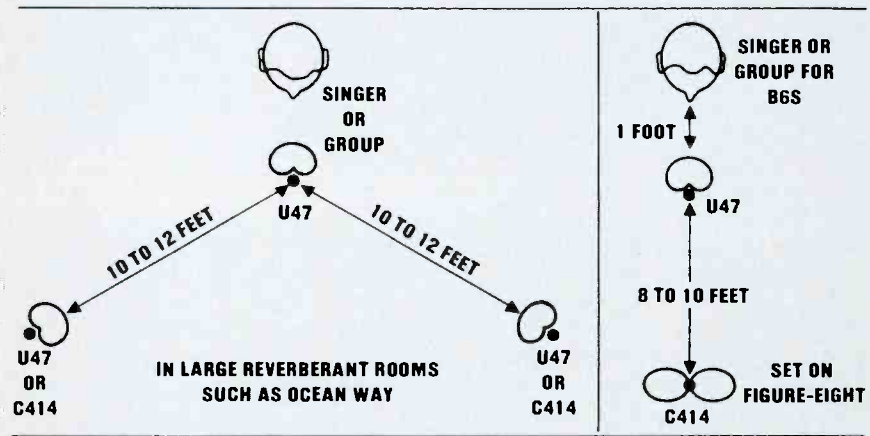
What about microphones, we queried? The common wisdom is that mikes are used for their specific sound. Is Eales comfortable with that approach? "Yeah, I think that's valid, because certain mikes give a complementary correction to distortions introduced by other parts of the chain and, in doing so, make things sound more real. Ideally, if I had absolutely transparent electronics, mikes that gave me what I wanted, and outboard gear to get the effects I wanted, I'd be in heaven! The whole problem is the gray area: you end up trying to balance one distortion against another."

M-S Vocal and Drum Microphone Techniques

"I like to use M-S techniques in the studio," he confides. "For example, when I record backgrounds, I use a center mike like everybody does, and then use a more distantly-placed figure-eight and split them. It's not *true* M-S, but you get a spatiality: it opens up a little window in the center for the vocalist.

"I orient the figure-eight microphone perpendicular to the cardioid pattern, and place it about 8 to 10 feet away from the singers [see accompanying diagram]. I use it in the M-S mode, summing it on one side and differencing it on the other.

"Another thing I do is use a cardioid and a pair of [Crown] PZMs, in-phase or out-of-



Ian Eales' microphone techniques for recording background vocals. The diagram shown left is appropriate for reverberant rooms, and involves adding the left mike to the left channel and right to the right (double) channel, about 6 to 15 dB below the main mike. The technique allows you to pan backgrounds in close to lead vocals, and still maintain "space" from it without resorting to the use of excessive echo on the background vocals. For the right-hand method, the 414 is added to one track and simultaneously to the other BG track, but set out-of-phase. Levels for both channels should be about 9 to 12 dB below the main mike.

JAY GRAYDON

tweak it out: they drive here from Channel probably six to eight times per mix. In a way, it's good for everybody, because they get fresh ears when they go back.

They get bugged at me because I really want to hear things in my own way. If I was mixing it myself, I'd actually make it *too* perfect, because I want to hear everything and I know *where* I want to hear it. This way, working with Ian and Verdick, I let the emotion come out more: I'll let lines jump out here and there if it's not a bad situation, because I'm hearing it fresh every five hours. I get a buzz sometimes from something jumping out. If I don't get a buzz, I tell them and they go back and do something about it.

R-e/p (RJ): Many people have commented that you get a very tight, punchy drum sound on your mixes . . .
JG: . . . As it turns out, last night we got the best drum sound I've ever heard.

R-e/p (ML): What track were you working on?
JG: A song for the movie *One of the Guys*, with Howard Hewett of Shalamar.

The snare drum was real ambient and, because it's gated, it isn't open when anything else is playing. But the drummer used Simmons toms. The foot's real "in your face," and the Simmons toms are "didged" to death. The sound of the whole kit is great but, because of the different sounds, you would have thought that it was a drum machine with an AMS [digital delay line sample] triggered on the snare. The sound is real unique, because it's a guy *playing* it — it feels so good.

R-e/p (ML): What did you mean by the expression that the Simmons toms were "didged?"
JG: Livened up with delay lines. We used two different delay lines so we'd have more of a stereo spread, Typically, the delay times will go from about 25 to about 50 or 60 milliseconds. We used a lot of feedback, but it wasn't "sproinging." We do that a lot with drum machine toms, just to give them some life.

R-e/p (ML): Did you choose the delay to correspond to the song's tempo, or was it used as just an effect?
JG: No, I don't time them up. I set the delay times to a point where the bottom-end doesn't get cancelled out; that's the key. We usually use a little bit of rocking — delay time shifting — so if it is in a cancellation area, it'll swim around. I never let it get to the point that it cancels all the way, and I may use it to build up the low-end.

R-e/p (RJ): Which sounds like a lot more processing than you might have employed on your earlier productions.

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For additional information circle #21

RAPPORT IN THE STUDIO — continued . . .

phase. Typically what I do is mount the PZMs on felt on the wall; the felt acts like a low filter. The close mikes are the predominant sound, with the room mikes used to get some air back into the track.

"With drums, this technique gives a nice punch and gets more of the image in focus. I find that it puts the drums in a better space: they're not 'right up in your face.' It gives you depth; somewhere to put the vocalist. Then you don't have to use as much artificial reverb, which tends to smear everything out.

"I think that we've been the gamut of using a totally dead room for tracking, and then adding artificial reverb. More and more people are getting back not to true stereo, but to using more room mikes, and live rooms. I try to make the echo feel like a room. In a live room, when you hit a snare, it's *loud*: it grabs your attention. In a dead room, it's a 'thud.'

"I have a record that was recorded in 1936 by Charlie Christian, the guitar player. It's mono, direct-to-acetate, probably two generations down now because it's on an LP. The horns have so much power, because the room is live. When I first heard it, I almost fell over when the horns came in. The *air* is what gives it the power. Unfortunately, there's a 'Way To Do Horns,' which is to stick a mike in front of each one. The way I'd like to do horns is get them all in the room, use a stereo pair, and have them play together: no doubling. That way, I could get that power."

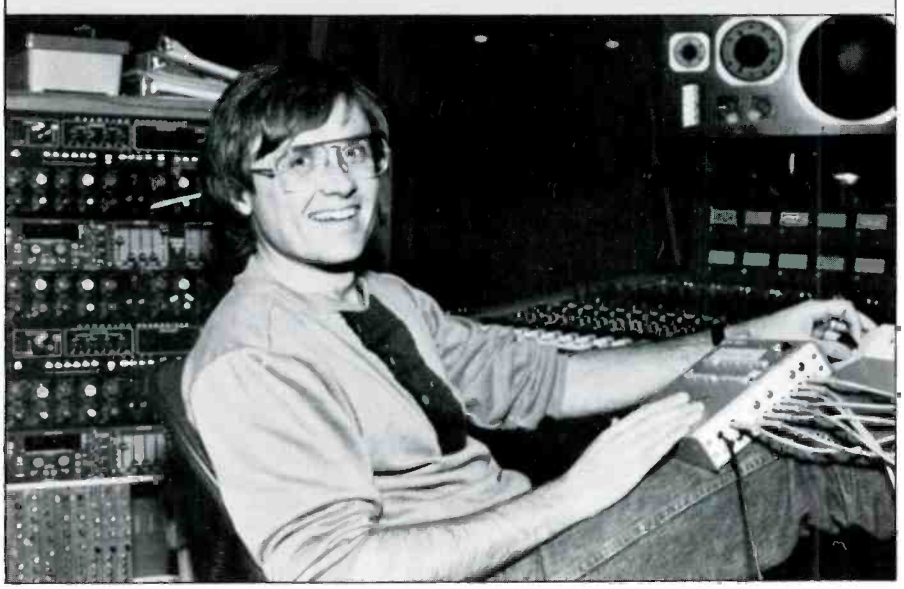
Of course, Eales works in the "real world," where horns usually are recorded one-at-a-time, and doubling is a way of life. How does he currently record horns? "I use Monster Cable microphone cable and come directly into the console, bypassing all the other studio wiring. The first time I tried this, I was using a [Neumann] U47 tube microphone on a sax overdub. I brought the fader up and reached for the EQ and, as the musician started to play, I *froze*. It was real: it had *power*, and there was no 'honk' like you often get with saxophones. It astounded me how dynamic it was! I'd used that microphone on sax before in that studio: the only difference was the cable.

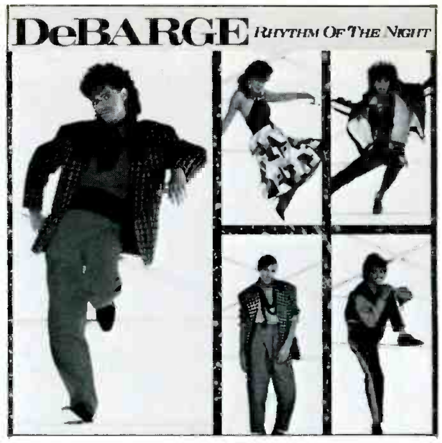
"When I mix, I use Monster Cable and go directly from the console to the tape machine, bypassing all the conventional wiring, as well as the console monitor electronics. I've built a custom volume control for my [Spica TC-50] monitors using selected, matched pots, and I use that in place of the console monitor section.

"Monster Cable mike cable removes a veil from the sound — the noise is taken back twice as far, and you can hear the echo out to the sides and behind. You get a much larger, more spacious soundfield. A lot of people like cables that are 'bright.' That's been a thing in our industry: 'Brighter is Better.' As far as I'm concerned, brighter is just more annoying!"

"The more I do this work, the more I want the equipment to be *musical*, and a lot of the equipment that we use in professional recording is *not* musical. In the high-end of consumer audio, there are some very musical components: Audio Research, Counterpoint, Monster Cable cables and cartridges, Infinity loudspeakers, the Spica TC-50 loudspeaker . . . all these names are probably foreign to most engineers.

"But there are those who are making highly musical equipment for the professional market — George Massenburg is one of them. His equalizer, in particular, is one of the best pieces of professional equipment ever made. And his computer is by far the best for automated mixing; it's light-years ahead of everybody else. If we could afford it, I'd like to buy one of George Massenburg's consoles. His have everything that I've wanted in a console, except the price! Just the other day, I played a tape that I had mixed on his console at his studio, and I was amazed at the sonic quality." □□□





JAY GRAYDON

Have you used similar prcessing on your latest work with Al Jarreau?

JG: Oh, yeah. First of all, it goes with the times. Also, Jarreau needs to move on; we can't keep making that same kind of record. If he's going to be competitive, we need to get fresh sounds. One of the effects that I did on this last record was on "Imagination." At the end of the intro, there's a falloff in the horns — they play a stab and, all of a sudden, the echo drops off in pitch. I sent that note to a [tape-delayed] chamber, and assigned it to a couple of tracks. As soon as that note hit, I grabbed the [tape-machine] VSO and sped the machine up as fast as I could. The illusion that I got was that the echo dropped off in pitch [because of Doppler shift]. I also used a lot of backwards echo on *High Crime* — backwards snare drum, horn and guitar echo — and I tuned drum machines in a strange way for percussion things.

There's a funny sound on "Tell Me" from Jarreau's record: the bass sound is a bottle! I knew what I wanted: a bottle-bass sound. So I found a bottle that was big enough, and filled it up with water until I got it right on A; I checked it with a strobe so that it was right on 440 cycles. I played a little flute when I was a kid, so I got the embochure across it, sampled it into a PPG Wave [digital synthesizer], and away we went. Robbie Buchanan played it on the keyboard.

R-e/p (RJ): You have an impressive collection of synthesizers and drum machines here at Garden Rake, and electronic instruments play a significant role in your productions. Do you use MIDI to hook them all together?

JG: MIDI was designed incorrectly — it should have been designed as a parallel interface, not serial. The crystal should have been one mHz, minimum. And all the synthesizers that weren't designed up front for MIDI should have had a *second* microprocessor for the MIDI interface alone, instead of taking up processing time in the current micro and slowing everything down. With some synthesizers, the delay between MIDI In and playing a note has got to be 200

JAY GRAYDON

milliseconds, maybe more. We all use MIDI; I'm not going to say that I don't, because I do. It's just that the delay times cause *major* problems.

I've learned a lot of little tricks for making drum machine tracks feel better, but they take a long time to discover. Even using the tricks, drum machines still have a mind of their own. They're each individual microprocessor-controlled pieces of equipment that time out the way they're going to time — they have their own little amount of "air" that they float around in.

On this last Jarreau record, I learned a lot about the way things feel. If you examine that album, you'll find that when I did use synths, it's pretty tight and feels good. Typically, I use a [Garfield Electronics] Doctor Click, after I've written the information into the synths and the drum machines. The synth is *always* going to feel later than the drum machine — just from microprocessor delays — so I'll fool around with the Doctor Click clock delay line, and tweak it around until it feels good. A millisecond can make a *big* difference in the way things feel.

R-e/p (ML): What main sequencer do you use on synthesizer tracks?

JG: I use the Oberheim DSX. The bass line on the sequencer is a control-voltage out, which is the way things *should* have been designed to begin with: CVs are *now*, gates are *now*. [Oberhiem designer] Marcus Reill designed his Expander module to have six CV and gate inputs, as well as MIDI. The gates and CVs take about a millisecond and a half off the sequencer before they hit, which is *fast*; MIDI is obviously much slower. When I'm doing bass lines, I write them into the sequencer using the OB-8. Then, when I'm printing it to tape, I pull the umbilical cord from the OB-8 to the sequencer. If I leave the OB-8 connected, the sequencer keeps looking to the synthesizer for interrupts; that lags the whole thing down and makes it feel terrible.

R-e/p (ML): You mentioned earlier that you prefer to mix at Michael Verdick's studio, one reason being that you liked the sound of his console. Can you elaborate?

JG: We were mixing the Jarreau record *High Crime* at Mike Verdick's studio, and his board is a Trident. He and Ian would bring the tape over here three times a day. Things sounded very different that they had when we mixed here at Garden Rake. I noticed that when they brought back the mixes, the bass drum had a much more defined center. It sounded smaller, more compact; it wasn't any less bottomy, though. Same with the snare — everything sat in a place better. So, I made an experiment with our console. I used a drum machine, and I listened to the foot on a non-VCA module, then on a VCA module — I matched the gains, so that wouldn't fake me out. I couldn't believe the difference. With the VCA module, it gets "soggy."

But we discovered that we can make these VCA cards sound better. Ian started changing chips in the modules. He wouldn't tell me where, but would put up stereo pairs. Ian liked the sound of a chip that costs $14 [an Analog Systems MA362 op-amp], and I liked the $4 one [an Analog Systems MA332]. It had nothing to do with price: Ian liked the "hi-fi" of the 362, and I liked the bottom-end of the 332, because it was tighter. I don't like the sound of 5534s anymore; to me, they just don't sound as good as some of these newer amps. I don't like anything that accentuates 3k and, to me, the 5534s hit a little harder in that area.

My whole objective in changing this amp is one thing: to get the bottom-end to have more definition, so that I don't have to worry about having other things sound too small. It's not affordable for me to buy another console, so we're going to go through the whole board and make it sound as good as we can. When I do buy another console, I'm going to buy the Massenburg [Moving Fader] automation, and another tape machine.

I don't like VCAs, and the reason is that they take up too much width in the track, especially on low-end sounds. The moving fader is more expensive, but it makes more sense sonically. Especially George Massenburg's devices. There's a VCA in his [Model 8100] limiter, and I still like it. The [Model 8200] equalizer and [Model 8500] mike pre-amp are also phenomenal. We have a lot of his equipment in our room, and I can't say enough good things about him.

R-e/p (ML): Do you normally use the Massenburg mike pre-amp modules, rather than the ones in your console?

JG: Very often. We use Monster Cable mike cable, bypass the studio mike lines, and go right to the Massenburg pre-amp, then to the Massenburg equalizer, and into the tape machine. We bypass all the other electronics every chance we get.

His is the *best* mike pre-amp in existence, as far as I'm concerned. I love the Massenburg pre-amp for guitar . . . drums . . . anything. But for vocal, it sounds *too* much like the guy in the room; it's *too* pure. I like it sounding more electronic.

R/e-p (RJ): If you're auditioning op-amps, it's pretty obvious thatyou're getting down to a very *fine level of sound quality. There are those who would argue that this degree of attention to sound quality is possibly pointless, because of what happens to the master tape after the mix.*

JG: I think that short wire *is* the way to go. The snare drum sounded much better with the Monster Cable direct to the Massenburg equipment, than it did going into a console module with about eight stages of amplification, taking the level up and down, flipping phase, and so on. If you've got the equipment to do it, why not just have it sound better? Monster Cable definitely makes a *major* difference in the way things sound. I was very impressed. I've A/B'ed it a number of times, and it's just unbelievable: a piece of wire, you know?

R-e/p (ML): Is it possible to hear the difference even with the signals passed through the console?

JG: Oh yeah, that's the first way I A/B'ed it. We were doing a background date, and we had the regular cable in. Ian went out real quick and changed it, and I couldn't believe the difference — it's just incredible how much of a difference wire makes. Then I got tuned into all the different wire that we had. Ian's thinking about going through the multitrack and rewiring it with Monster Cable.

If I didn't have the equipment in my own studio, I wouldn't do it. But I have the equipment, so why not do it? It's going to sound better. The mix never sold a record — if the song's not there, what's the difference? But we *have* the equipment, so let's do it — it sounds better for us.

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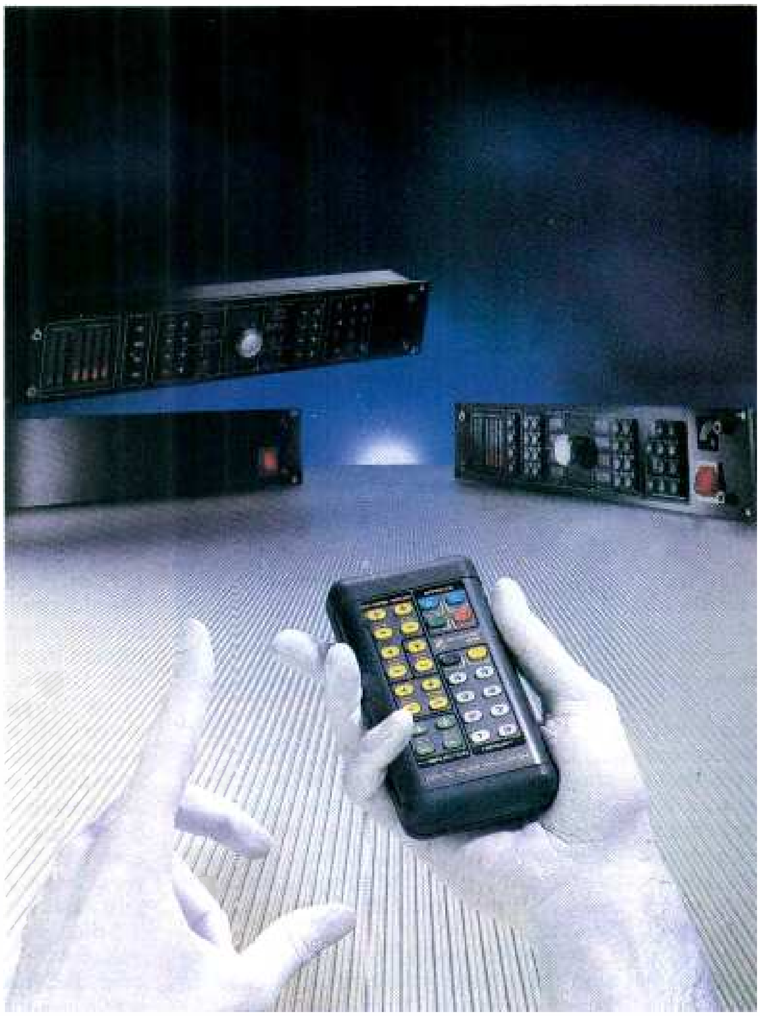

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Density of resonance	Max 3 per Hz of bandwidth depending on room size
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For additional Information circle #23

JAY GRAYDON

The sonics of a record are not going to stop me from listening, or make me listen to it one way or the other. I think about sonics just for a brief moment: I'm listening to the record.

R-e/p (RJ): But you go to great lengths in your equipment selection, and seem to really care about the sonic detail of your own work.
JG: That's basically for my high standards — for me. I'm not trying to impress anybody with it. I'm just trying to make a record, and I want it to sound as good as I possibly can.

R-e/p (ML): So a studio, for you, is like a tool: if you were a woodworker, then you'd have sharp chisels instead of blunt ones, because they would make your job easier?
JG: Of course. The bottom line's the *music*.

R-e/p (ML): What monitor speakers do you use at Garden Rake?
JG: Yamaha NS 1000s. I use Auratones occasionally, just to see how things are sitting on a small speaker. I never use the large monitors in the Garden Rake control room. You can only have *one* set of monitors in a room that are in the right perspective. I don't care what anybody says, there's only *one* correct spot for the monitors. Everything else, unless its higher or lower, is just going to be in the way.

R-e/p (ML): If you had a free choice on your next project, which artist or band would you like to work with? What do you think you could do for them as a producer?
JG: I want to work with Shalamar. Howard [Hewett, the goup's leader and tenor vocalist] and I are talking about working together with the group. We just did a tune for a movie called *One of the Guys* — it'll be out shortly — with Shalamar billed as the artist. But I don't know how the entire band plays — and I didn't have the time to find out — so I had to use a studio band with Howard, who sang the tune. I let the track be a little loose — just waited until I got a good-feeling take, instead of worrying about precision — and wailed through the vocal because the guy's a great singer.

What I can do for Howard is to take the "funk attitude" more into a rock and roll attitude, because he has the ability to sing rock and roll. Obviously, I want to reach as many people as we can — anybody in this business wants to have the crossover point. Now, because of Madonna, Hall and Oates, and other people, funk and rock and roll have met, and I think it can go even more rock and roll, with the funk attitude still there.

My favorite singer in the world is Stevie Wonder; my total musical idol. I've worked with him a few times, but the guy doesn't need any help! Unfortunately, most of the people that I want to work with don't need me. [Laughter]

I talked to Chaka Khan the other day; she was over here doing some tracks. I think she's the best female singer on the *planet*. If I were to work with her, I'd fill up as many tracks as I had, just so I could pick and choose her licks, and turn them into little masterpieces.

I was very impressed when I heard Wham. Their concept's right — I just think it should be a little tighter. To me, the kind of songs they're doing are "classy" songs. For example, their new tune's got a disco hi-hat on it. Seven years ago, that was dead: nobody could get away with it but somebody like them. Now, that's daring, and I'm glad they got away with it. I just think their material could be a little tighter. It would be a little more "popping," a bit more "snappy." When things get loose, they lose a little bit of punch. I think the singer's great, and I think that he could use that.

"I love to play jazz, but I *don't* want to make jazz records."

Typically, bands that are happening don't fire their producer. If they're on their way down, that's the time they want to make a change. I'd almost rather do a new act than a band on their way down, because when they're on their way down they're nobody's friends anymore! It's the old story: you're only as good as your last three minutes. I'm going to look for new acts that I think I can develop.

R-e/p (RJ): It sounds as if you're looking for someone whom you can direct fairly specifically in the studio.
JG: I have a tendency to do that. Most of the direction comes from me forming the tune from the bottom up. If I'm the writer, I'm really in good shape, because I've got so much time with the tune that I really know what I want.

R-e/p (ML): You are a songwriter, a session guitarist, and a major producer. When you hear a basic demo that hasn't got any of the "modern sounds," how do you tell that through the multitrack process it can be turned into a hit record?
JG: In the old days, I wanted to hear the *simplest* demo possible, because I didn't want to be influenced by the arrangement. Now, I do. If somebody comes up with a great drum-machine pattern, or some sampled things, I now want to be influenced by that texture.

R-e/p (ML): So you want ideas from the band. But what new band can afford the cost of a complex sound on a demo, especially when the "industry wisdom" is: "Keep it simple; just give me a basic drum sound and bare tracks?"
JG: First of all, if it's a band they're going to have their material pretty organized. If it's a rock and roll band, there's not that much to do. Let's take a crank band . . . Van Halen for example. Van Halen is *Van Halen*; there's not a lot you're going to do with them — it's a crank trio. If I were to produce them, I'd just go at making the attitude of their tunes great.

R-e/p (RJ): What about developing a solo artist from the ground up? You're an excellant writer and arranger, and you have access to good material. What about finding someone who is just a voice?
JG: A solo artist from the bottom up is a good move for me. I wanted to produce Mike McDonald: he ended up doing it himself. I thought I could help, but he should produce himself if that's what he's comfortable with. I met with a guy named Robbie Neville — I didn't get the job. He's a songwriter who is of the "now" songwriters, and he's a great singer. My ego doesn't get blown out — he made his right move. Out of it I got a friendship with a guy that I like a lot, and a songwriting partner to work with. So, the gigs come and the gigs go. If the artist wants to work with a guy, and the record company wants to work with a guy, that's going to be what happens.

I did an album with Donny Osmond, called *4-1-1*. It never got signed, however, because Donny said in print that we were going to sign with Qwest. Quincy [Jones, Qwest president] didn't want to let that fact get out of the bag, because we didn't want anybody to know this was Donny Osmond coming in. If you had heard this record, you wouldn't *believe* how good a singer he is. But after the Quincy problem, nobody wanted to know about the record, because . . . why didn't we end up on Qwest? Now, we're poison. The record is in the can, and it probably will never be heard. It's a shame, because he guy is *so* good. I thought I was going to be the hero: taking somebody that's got such a reversed image . . . but the business people just didn't want to know about it. So, there it goes.

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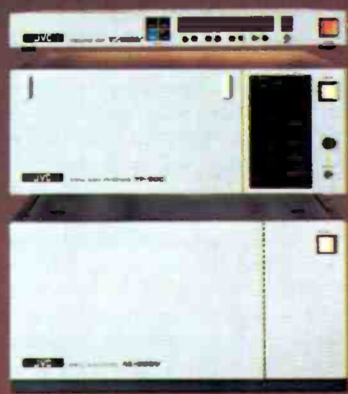

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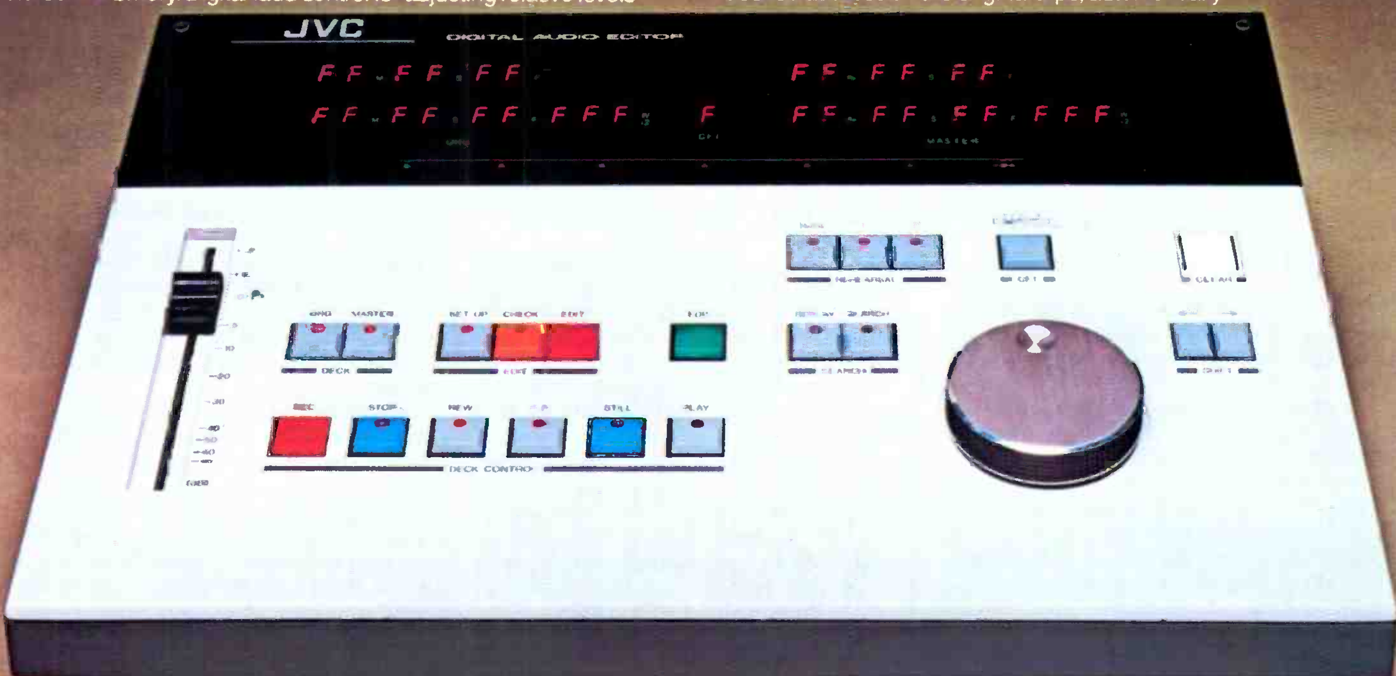

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JAY GRAYDON

this: the record company really has to be behind a new artist; it's got to be a *major* deal. It has got to be somebody that the label is *really* going to go with before I will do them, for one reason: otherwise I'll work hard and try to make a great record that ends up on the shelf! Now, when I hear a new artist that they might be interested in me doing, I investigate the finances of this artist. How much are they giving them? What's the budget? Are they really into this band?

These days, you cannot break a new band without a mega-dollar independent promotion budget. If the label doesn't like the album, that's another story. But I know that they're going to like what I'm going to give them, because I make records well. I know my job — I've done it a long time. I'm tuned into the radio. What I give them will be workable, and there will be songs for them to work as singles. So, it's just a matter of establishing whether they're going to do it or not, and I've *got* to know that before I go in.

R-e/p (ML): Does the record company give you complete control over the final product?

JG: It depends on what record company it is. My home label is Warner Bros. I'm not a staff producer; I'm independent. But I basically deal with Warner Bros, because I like the way they work. With the Jarreau record, we had cut all these songs. We listened to them collectively at one point, and realized that some of the material was not going the way we wanted it to. And I agreed with them. But now we need additional money to record some more songs. Obviously, Jarreau is a big act, and we got the money. We recorded some more songs, and ended up with a record that I think is the best one we did of all. We haven't popped a single off it yet, which is trouble. But, at this point in his career, Jarreau is not Duran Duran; he's not Michael Jackson. He's a sophisticated artist, and it's hard to get him to do tunes that are lower IQ for the radio.

R-e/p (RJ): It would appear that one of the primary considerations in getting a good performance out of a singer is that the material has to be right for them.

JG: Material is the *whole* game: if you don't have good songs, you've got nothing! You know, there are very few times that artists make it on momentum. Although the momentum of an artist can carry bad tunes sometimes, it doesn't happen often. And a bad tune will *never* break a new artist; you've got to have hits.

R-e/p (ML): What makes for a good song: lyrics, the melody, or the feel?

JG: Everything — I think it's all one. Today, more than ever, the way a record is made is affecting the market. For example, "Owner of a Lonely Heart" by Yes was the first major sampling tune that we heard, and I thought that was a monumental record — I thought it was a good tune too. Thomas Dolby's "Blinded By Science:" a great record. The tune was okay, but it was a *great* record. "Shock The Monkey" by Peter Gabriel: that was the first time I heard Simmons drums. Now when you hear the song, it sounds dated — it's only been two years since it was released, and sonics have changed.

A good record can hold up a fair song. Space is also important, which is why I say that good tracks are a key to the game. If you notice, most hits are not crowded — they usually have a lot of air in them, and they're real simple; easy on the ears. Take the tune "Tell Me" on Jarreau's record: I'm bucking for a single. Nobody else thinks it's a single, maybe because it's a little sweet sounding. But I'll tell you, it's so hooky and so open!

I like "low-IQ" songs; I like tunes with a lot of air that are very simple. I don't care about, "Dig how tricky we are with these chord changes. And dig how cool this melody is that winds through the changes." I've done that, man. I want to get to the people, and I can still make real good records that way.

R-e/p (RJ): The artists that you've worked with up to now seem to fall into a fairly defined range of musical style, basically Jazz/R&B Crossover. Yet your songwriting credits reflect a much broader range. What has governed your choice of artists to produce?

JG: There are a lot of gigs I turn down because I don't want to stay in that *bag*-thing that I'm in: guys that are jazz R&B, and who are trying to get to pop. I don't want to know about jazz anymore!

I love to play jazz, but I *don't* want to make jazz records. And not just for the money reasons. Look, we *all* want to make money, but I want to get to people. God put me on this earth to make people happy through music. I'm doing something for human beings; I'm making people happy by making records. But you've got to look at the logistics, because they go along with it. Obviously, if they are getting to a lot of people, they sell, and they make money for everybody. The first concern is the record company, because this is the music *business*. So, if I'm not selling records, I'm not doing my job.

But I'm always going to make money in this industry: I'm not worried about eating. My basic thing is to make people happy. I'm doing something that hardly anybody in the world gets to do. When somebody's making a box of Kleenex, for example, they are not making anybody happy: they're just giving them a useful product. But when people turn on a song, they might say, "Oh yeah, Mary and I were sitting there the first time I heard that tune." It's fond memories — when they turn the record on they get a good feeling. They're driving home in the car with the freeway stopped, and they're bummed out. But dig what I'm doing for them. A tune comes on, and all of a sudden they're grooving. Boy, that's a great feeling. When I was a studio player I didn't think like that.

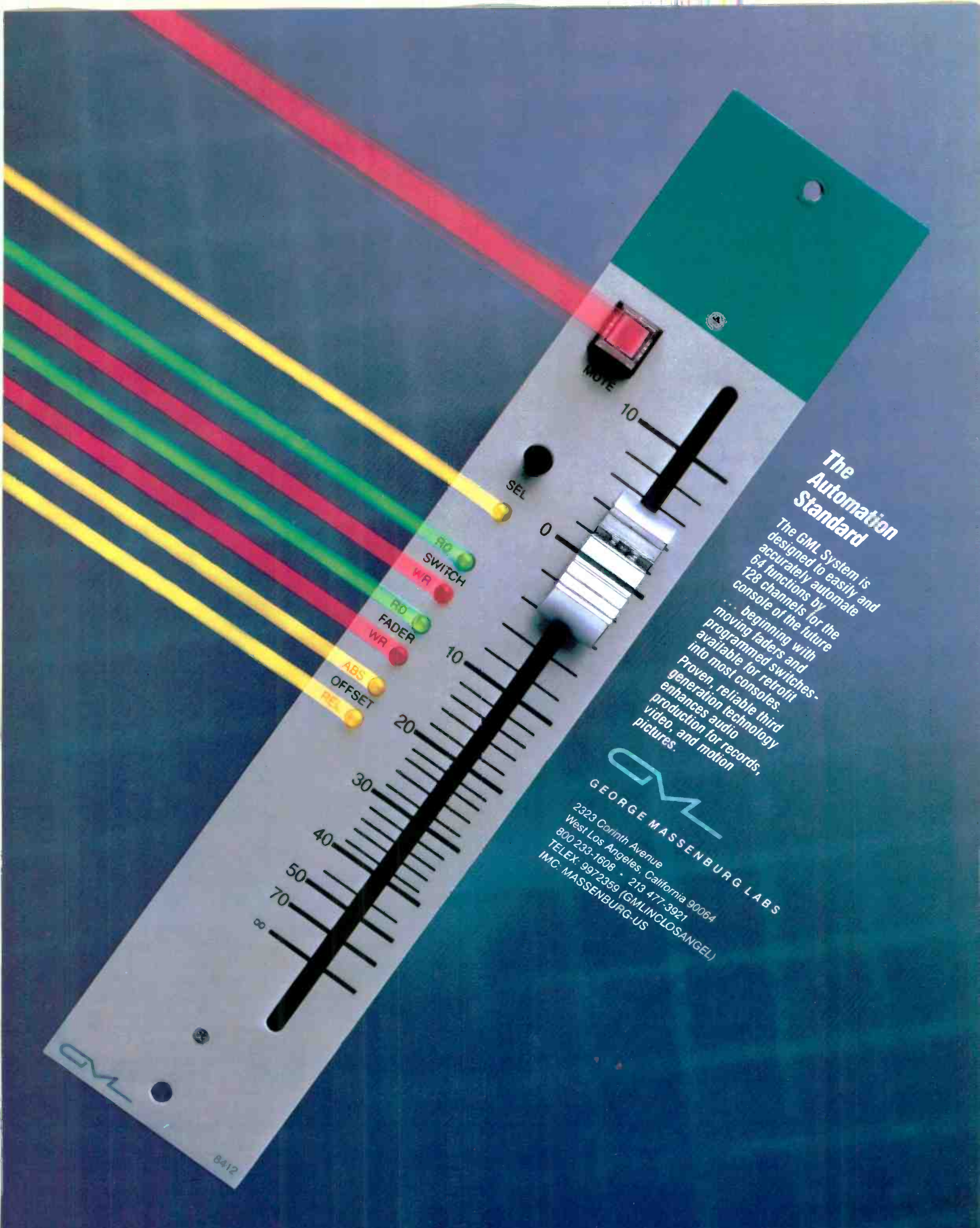
With the job that I have now, through all the Hollywood bullshit that goes down, no matter what they try to do to me — take away from me businesswise, mess with me . . . whatever — my final result is that I've made somebody happy. That's the ultimate.

R-e/p (ML): When you are working with session musicians of the caliber of Michael Omartian, David Foster and Robbie Buchanan, do you find that you get on with them more easily as a producer, because you're of equal status?

JG: Yeah. See, the younger guys that are coming up don't have the experience — I don't have the time to train them. And anyway it's going to take time to get what I'm looking for. It's hard to get guys like Omartian, because he is a producer, too. Every once in a while, I can get Michael on a lull. David Foster's my best friend, so we do things for each other; it's a give-and-take thing. I don't use the B-string players — the guys that are coming up — until they're A-string players.

Guys like Omartian want to get in and out of a session real quick. All these guys cop attitudes on me. Because, first of all, we're friends. I mean, I get attitudes from these guys all day long! [Laughter] Just like I'd give them if I was on their date; *I* don't want to sit around making it 80 times. I put on my studio hat, and I want out when I walk in the door. But, seriously, it's not really that bad: we help each other out. They don't care about the reasons, but they'll play it again.

Now making a record . . . I come up with a great tune or do a great job on a record and I go, "Whoa, yeah. That was *nice*. I brought it off!" ■■■



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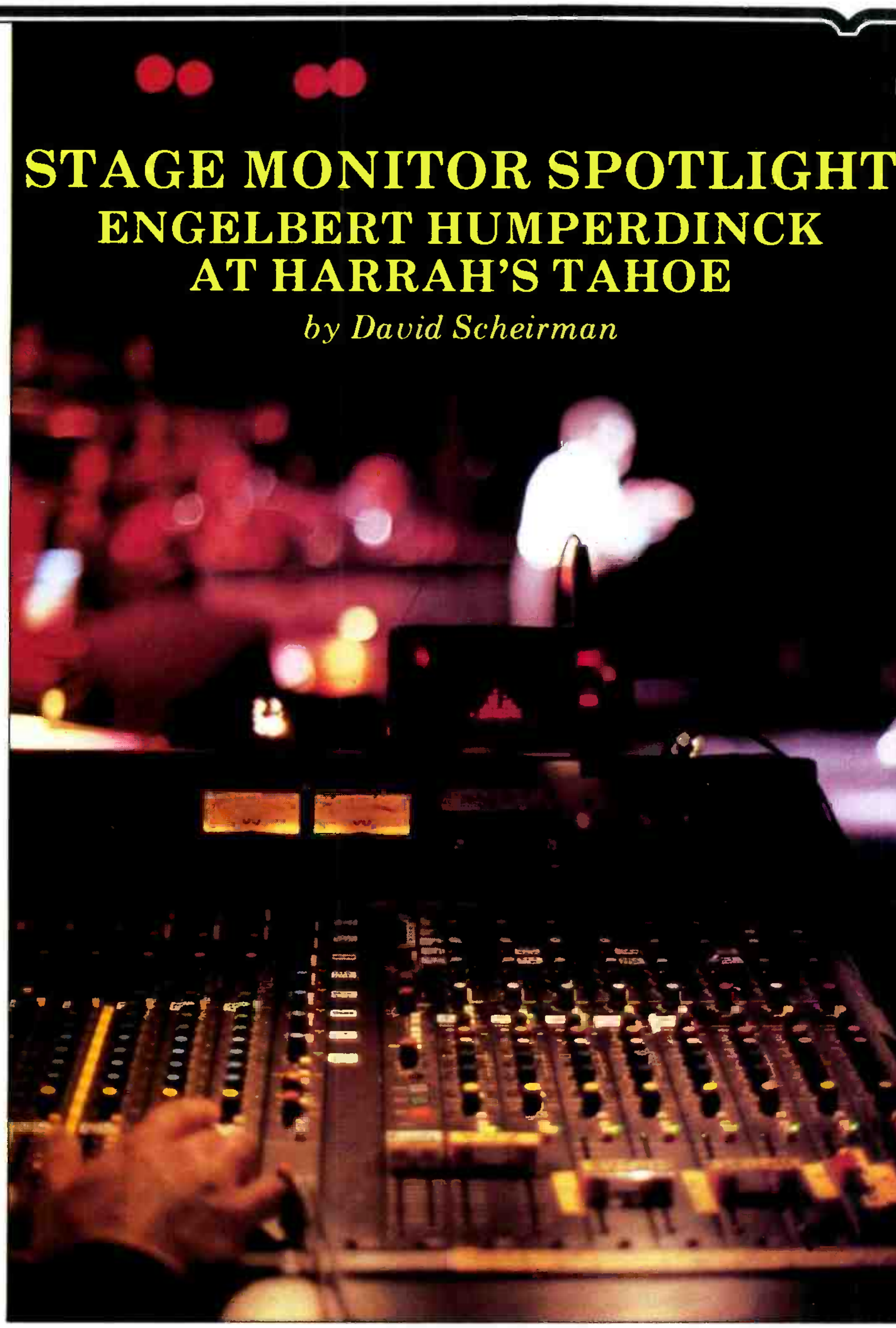
STAGE MONITOR SPOTLIGHT
ENGELBERT HUMPERDINCK AT HARRAH'S TAHOE

by David Scheirman

Within the past 10 or so years, stage-monitoring systems for live-performance use have become increasingly complex. As various manufacturers have responded to the mounting market demand for dedicated stage-monitoring consoles and loudspeakers, a versatile array of equipment has become available to sound reinforcement companies involved in the assembly of such systems.

Once serving only the exclusive domain of touring rock acts, concert stage-monitor systems have been expanded in scope to include a wide variety of entertainment events. Sophisticated systems may be seen these days serving such diverse activities as theatrical events, worship services, Broadway productions, and resort hotel shows.

Recently this writer had the chance to try out the A-1 Audio stage monitor system that has been assembled for use by popular entertainer Engelbert Humperdinck, a singer whose stringent sound requirements have put a host of sound reinforcement firms through their paces during the past several years. I heard the system and observed the show at Harrah's Tahoe, a resort hotel and casino located in the mountains of Nevada, in November 1984.



All photography by David Scheirman

System History

A-1 Audio, a company with offices in Atlantic City, Hollywood, Las Vegas and Lake Tahoe, has specialized in serving the casino show market, and over the years, has developed a method of serving the entertainers that frequent this circuit with a good measure of success

"Each sound system that is leased to a client is tailored to suit his or her particular show," explains A-1 owner Al Siniscal. "Different shows will require different types of gear — a certain console, perhaps, or a unique type of loudspeaker. Upon observing an artist's show, certain concepts will be tried out, and combined into the sound design that we feel will best suit that show. However, over and above the type of hardware used, the particular personalities of the system operators can be *the* most positive aspect of the system we send out to service a performer."

Engelbert Humperdinck's show has seen many personalities come and go, as well as many different sound systems; since early 1983, a total of 18 different individuals operated a variety of stage monitor systems. The system provided by A-1 Audio — which seems to have finally provided that entertainer with the sound he has been looking for — relies heavily upon loudspeakers manufactured by Meyer Sound Laboratories. The individuals who made the show's sound happen for the 1984 Fall tour included Edward J. Spitzig (monitor mixer) and David Dansky (house mixer).

Artist Rapport and Communications

Humperdinck has developed a set of hand signals with which he communicates his on-stage audio needs to Spitzig. "Perhaps of the greatest difficulties one this artist has had in the past was finding an individual who could correctly interpret his expressed needs," notes Spitzig. "Being able to have that immediate constant communication during the show is crucial to the success of each show."

Lighting director George Boyd concurs: "With this production, the on-stage sound is absolutely *the* most important thing. We always shoot for the best, of course, with the other production systems such as lighting, projection and staging. But the monitors have to be right, every night."

Spitzig maintains verbal communication with the performance area and the house mixing position via a Clear-Com intercom system (Figure 1). Handsets are located on stage at the conductors' and background singers' positions. Both house and monitor consoles are equipped with handsets, signal lights and sonalert (audible tone) devices. In addition, a

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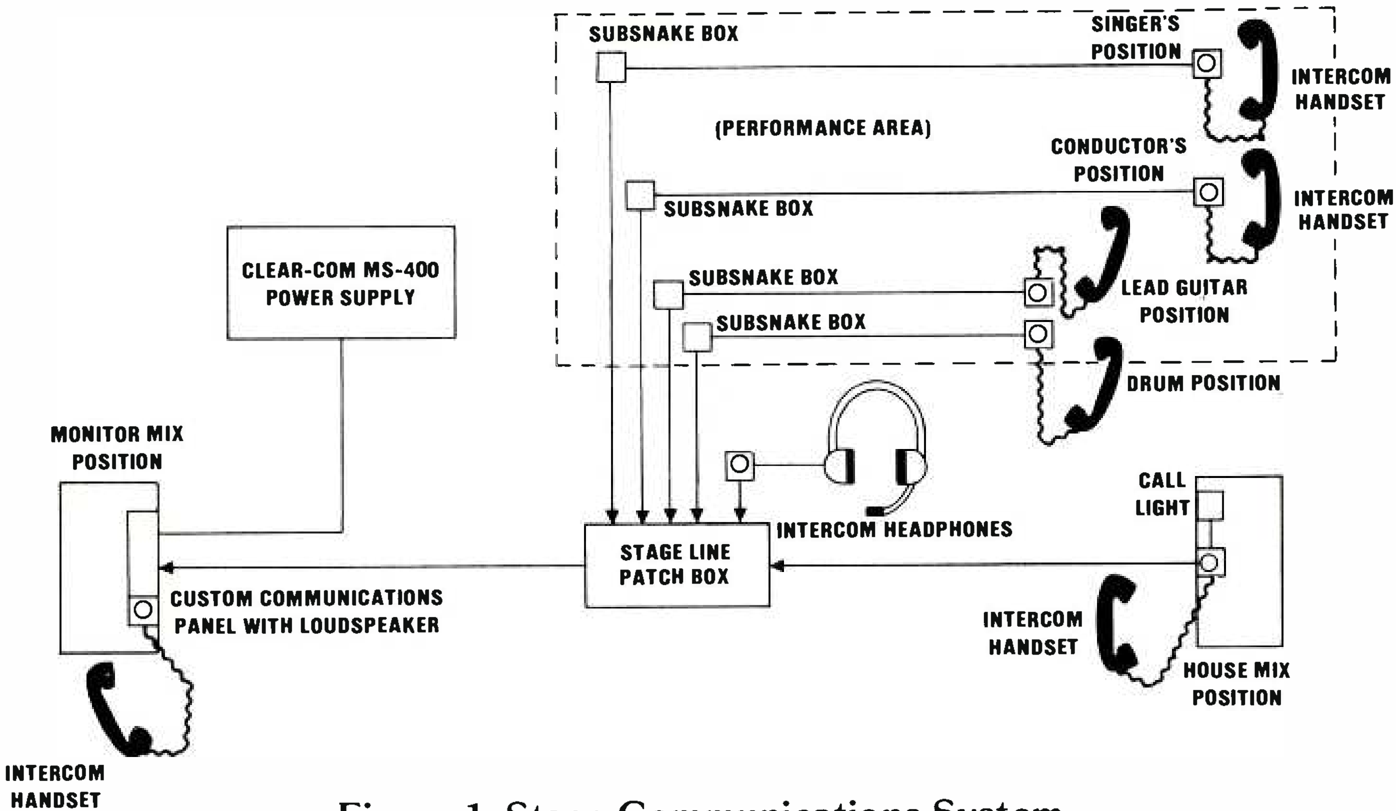

Figure 1: Stage Communications System

Englebert Customized STAGE MONITORING

Beyer headset with microphone is located at the stage mike line patching and termination point behind the stage set.

A Soundcraft 800B monitor console has been modified by A-1 Audio to include an on-board communications panel that houses a miniature two-inch loudspeaker, which may be switched into the communications line. "When the house engineer needs to talk to me during the show, I have the ability to hear him speaking to me directly from the speaker on my console, without having to take my eyes off the artist or my hands away from the board," explains Spitzig (*pictured left*). "This way, he can notify me of certain frequency overtones that may be rolling off stage from the monitor system."

A Clear-Com MS-400 power supply and master station energizes the communications system. One audio line in the multipair snake system is dedicated to the intercom signal.

Monitor Mix "Zones"

A specific fix included in the sound design for Humperdinck's show includes the concept of monitor mixing "zones," since the performer travels to different parts of the performance area. To blanket the entire stage area with the required high-level vocal reinforcement mix would cause a serious leakage problem for the microphones dedicated to the horn, string and percussion sections of the orchestra.

To counteract this potential problem, stage monitors for just Humperdinck have been set up on seven discrete output mixes (Figure 2). Three mixes handle downstage center, left and right, and sidefill cabinets receive stereo fourth and fifth mixes. A midstage performance deck, served by two cabinets, receives a separate mix, while the upstage show intro platform and stairs require yet another mix.

"A variety of methods were considered when the stage system was first put together for this show," monitor mixer Spitzig explains. "Overhead monitors would give us good coverage, but the speakers would be too far away from the performer to deliver the high-quality, 'near-field' sound levels that are required. Sidefill cabinets cannot be aimed back towards the orchestral sections, or we would have a tremendous problem with 'leakage' into the open mikes, particularly in the string section. Since Engelbert does travel to different areas during the show, I fade the various mixes in and out so as to give him the best coverage at each location [Figure 3]."

Positioned along the downstage edge of the performing area, six Meyer UM-1 Ultramonitor™ cabinets are fed from three separate mixes: center, left and right. Each pair of cabinets is driven with a Meyer M-1 signal processor and a BGW Model 750B power amplifier. A Yamaha Q1027 third-octave graphic equalizer is patched into each output mix from the Soundcraft 800B 32-input console; Figure 4 details the full monitor system signal paths.

A pair of Meyer MSL-3 cabinets are used for sidefills (Figure 5).

"I find that it takes a bit of juggling between the sidefill and floor slant mixes to obtain the perfect combination of 'near-field' and 'far-field' sound energy," states Spitzig. "If I rely on the front slants too much, the stage sound is not 'open' enough. Yet, the downstage mixes have to be hot enough to give the performer a sense of immediate, 'right-there' sort of presence."

A pair of Meyer UPA-A1 cabinets cover the deck area of the stage set, a narrow carpeted walkway approximately 36 inches above the stage floor. "As the performer enters the deck area from the stairs in front of it or above and behind it, I carefully fade that mix in while lowering the level of the speakers covering the area

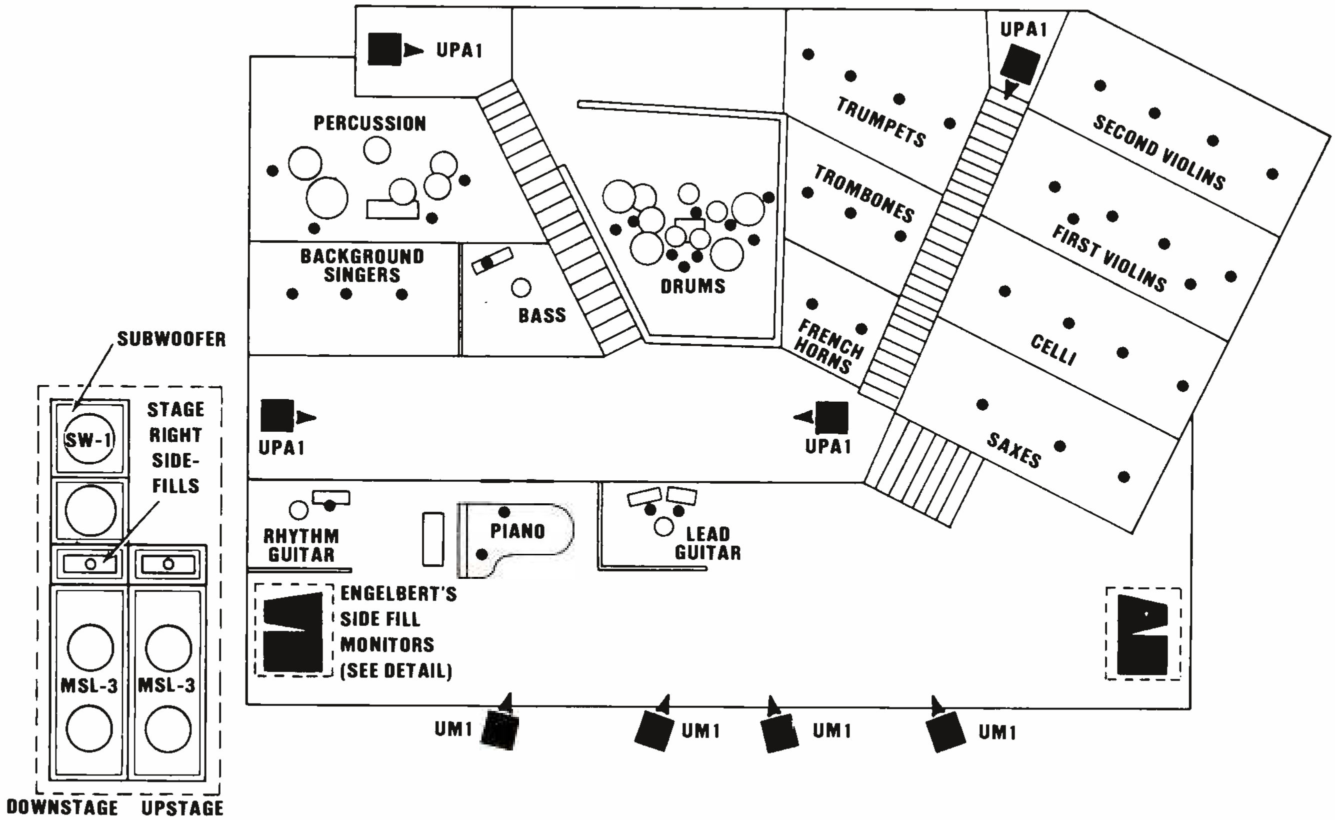

Figure 2: Engelbert Humperdinck's Monitor Zones

Englebert Customized STAGE MONITORING

from which he just came," Spitzig continues. "Then, he has additional cabinets positioned atop the stairways that lead to the upstage part of the set. Each time he makes a trip back there, I have another mix to rely on."

In addition to Humperdinck's seven monitor mixes, there are five other mixes serving the stage area. Monitor cabinets used for these mixes include a Yamaha Model S2115H floor slant for the drummer's position; two Community Light & Sound NC-12 small slants for the bassist and lead trumpet player; a pair of JBL Model 4602 slants for use in the deck area by the three female background vocalists; and three M&K (Miller & Kreisel) Satellite Speakers in the girls' primary performing area.

"The M & K's are very small and very brilliant, so they work well in this application," notes A-1's David Dansky (*pictured right*). "However, they are not as efficient as most pro-sound loudspeaker units, having originally been designed for home hi-fi use; they soak up all the power that a BGW 750 can give them. Consequently, we are experimenting with other small cabinets for the background vocalists, including Anchor monitors and a new product from Bose [Figure 6]."

Frank Leone, the show's pianist and musical director, is provided with a Yamaha MS-10, self-powered monitor system, which offers him a reinforced signal from the Helpinstill piano pickup (Figure 7).

Figure 3: A Meyer UPA-1A speaker cabinet serves one end of the "deck" monitor zone. The mixes to various zones are faded in and out as the performer moves to various parts of the stage set.

Guitarists located in the downstage area do not require separate monitor speakers, due to their proximity to the featured vocalist's sound field. No reinforced sound is fed into the orchestral chair sections, excepting a Community Light & Sound NC-12 cabinet provided for the lead trumpet player.

Instrumentation and Stage Mikes

Achieving a consistent stage (and house) mix in varied acoustical environments requires a rather complete group of console inputs. To this end, no area microphones are used for the various orchestral sections; instead, each individual musician receives a separate microphone, even in the smallest of venues.

A total of 56 mike lines reach the monitor mix position (Figure 8). Here at "control central," four Yamaha M406 rack-mount units are used as submixers to drop the total number of inputs actually reaching the Soundcraft 800B console to 32. Submixed groups include percussion instruments, trumpets, French horns and trombones, flutes and saxophones, violins, cellos, and background vocal mikes (Figure 9).

"The M406 is a versatile tool," notes Spitzig. "Each unit has a left and right output, as well as an auxiliary out. The same mixer can be used to do three separate group by using the

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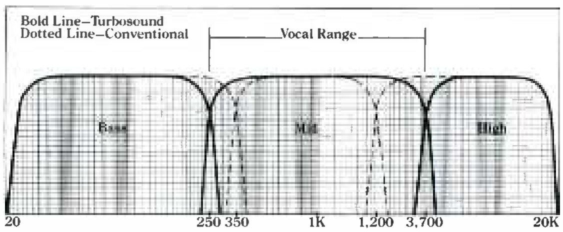

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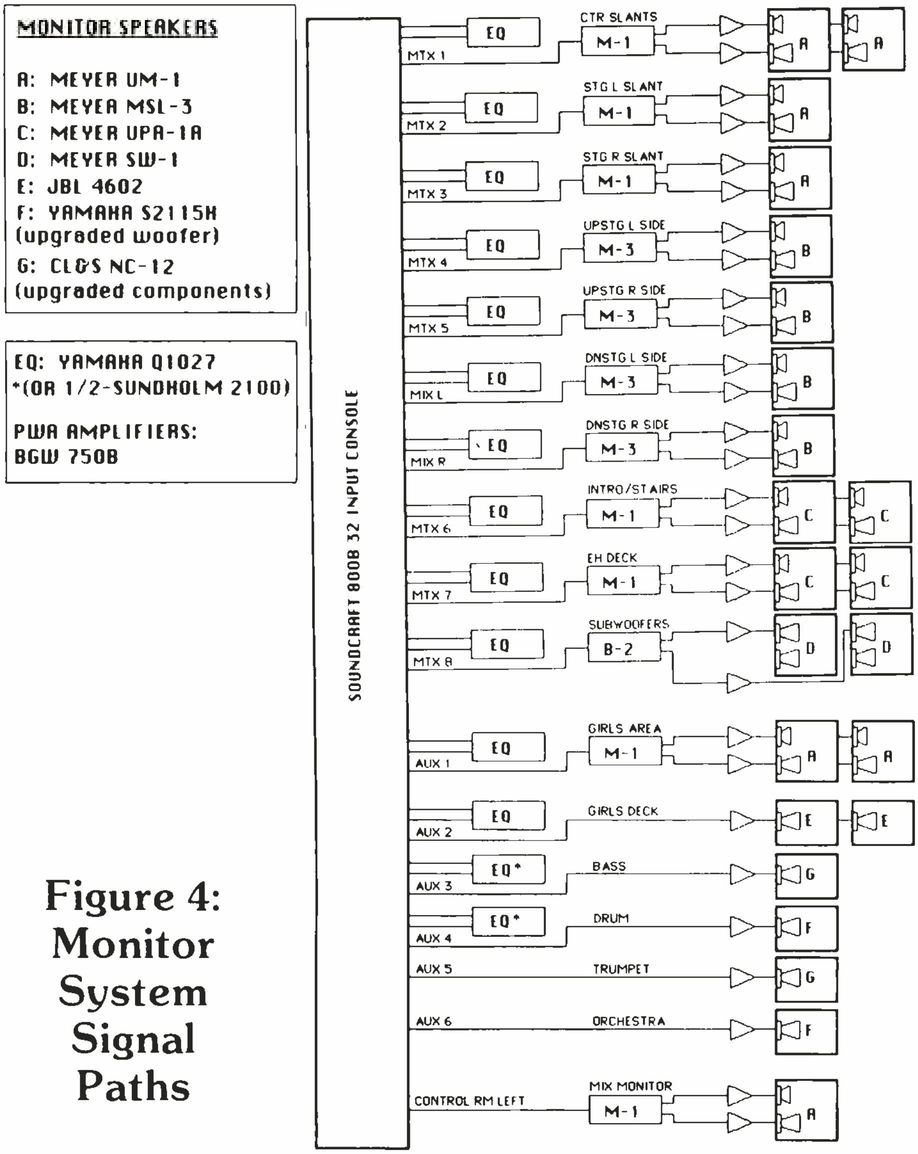

Figure 4: Monitor System Signal Paths

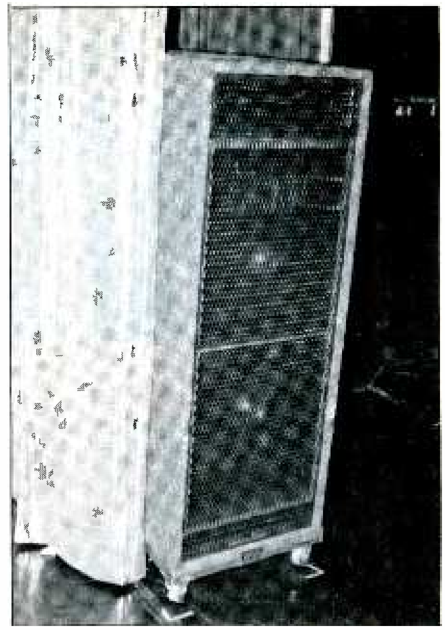

Figure 5: One of the two Meyer MSL-3 speaker cabinets used as sidefill monitors

panpots."

Vocal mikes used for this show include gold-plated Shure SM-87s for the featured vocal channel and a spare line. "We keep a variety of microphones on hand for Engelbert," Spitzig notes. " We are currently using the SM-87, but Electrovoice PL-77s and an AKG D33OBT are occasionally called for." Background vocalists receive Shure SM-78s.

Shure SM-57s and Sennheiser 421s are used to pick up the brass and reed sections, while violins and cellos play into Shure SM-17s and SM-18s. "These small microphone elements are wrapped up in a foam 'mouse'," David Dansky explains. "We have found that the foam will protect the valuable string instruments from damage. It is a method that works much better than the old clip-on brackets commonly used a few years ago."

Additional inputs to the monitor console include the left and right sides of EXR Model EX-IV Exciter; a stereo return from a Lexicon 224X digital reverb unit; and a direct feed from a Helpinstill pickup installed on the grand piano.

A Yamaha R-1000 digital reverb unit processes the violin submix. "Strings can sound really naked when amplified electronically," Spitzig notes. "A taste of reverberation helpens to sweeten the string mix. The R-1000 sounds really clean, and is a good, compact, rack-mountable unit."

Setting up the Monitor System

"With this many mixes passing essentially the same vocal input, but different program material, care must be taken to adjust each one separately before trying to check the sound of the whole stage," says Spitzig of Humperdinck's stage monitors. "I will us-

Figure 6: Three M&K (Miller and Kreisel) high-fidelity Satellite loudspeaker units serve the background vocalists.

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The H969 is an addition to Eventide's full line of Harmonizer special effects units. Our industry standard H949 is still going strong. The H969 ProPitch Harmonizer maintains Eventide's leadership position. For your most demanding applications, the H969 represents the state-of-the-art in pitch change technology. Hear it at your Eventide dealer soon.

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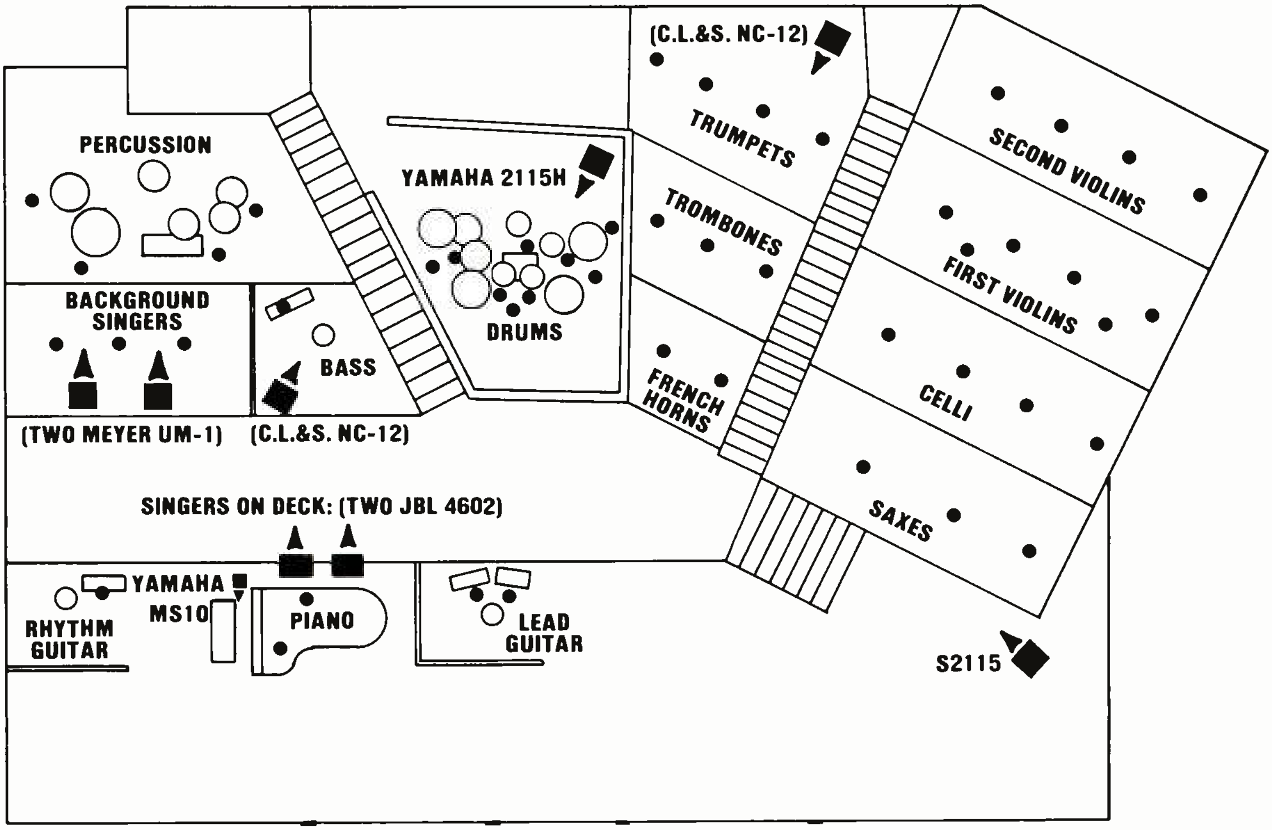


MUSICIANS' MONITOR LAYOUT

ACOUSTIC MASKING IN STAGE MONITORS
What It Is and How to Eliminate the Effect

by Tracy Crawford, design engineer Klipsch and Associates, Inc.

One of the primary needs of every performing musician is the ability to clearly hear himself or his instrument on stage. In order to do this, a stage monitor must overcome the masking effects of the PA stacks or house sound system.

Masking is a situation in which the ability to hear one sound is reduced by the presence of another. The nature of this psycho-acoustic phenomenon is such that a low-frequency tone will mask a high-frequency tone. For our purposes, masking can be divided into two general categories: bass masking, and equivalent-frequency masking. Bass masking occurs when a low-frequency sound masks a significantly higher frequency sound, while equivalent-frequency masking occurs when the masking sound is similar in frequency to that of the masked sound.

To get a feel for the effects of bass masking, we conducted a non-scientific test in which an equalizer was used to add about a 5 dB boost, extending from 40 Hz to 200 Hz, to a loudspeaker that had a very flat frequency response. We had several people listen to the speaker while several cuts of music were played. The result was agreement from everyone that there was less detail and clarity when the added EQ was in the circuit. Expressions like "muddy," "mushy," "interference," and "run together" were used to describe the "EQ-in" condition. One person described the sensation as one of having a wall placed between himself and the speaker.

This lack of detail was also evident in a stage monitor we recently tested. Upon further investigation of the unit, we discovered that the level from about 50 Hz to 500 Hz was 2 dB higher than the midband. All of which suggests that a small amount of additional bass can contribute to serious difficulties in hearing detail, and masking studies have clearly shown this to be true.

The other form of masking — equivalent-frequency — is the most common, although it is rarely thought of in terms of masking. Figure 1 gives an approximation of the masking ability of a 500-Hz tone as a function of frequency; the curve is similar for other frequencies, and for bands of noise. It is easy to see that the degree of masking is greatest near the frequency of the masking tone, which explains why one occasionally has great difficulty in following a conversation in a crowded, noisy room. This phenomenon also illustrates why musicians have problems hearing themselves on stage.

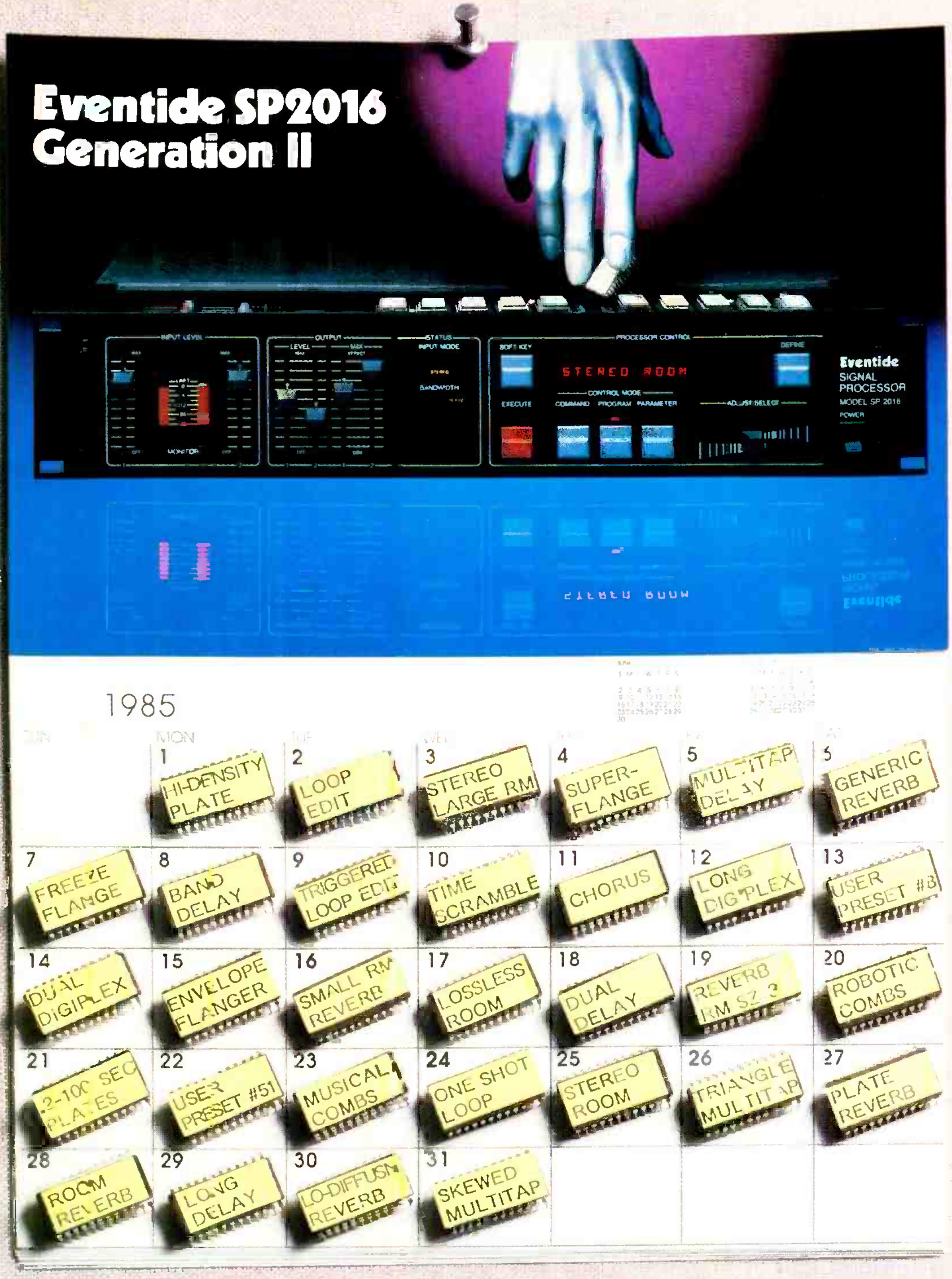
Generally speaking, all monitoring problems fall into one or both of the previously discussed categories. The solutions are relatively simple in theory, but perhaps a bit more

ually start with the downstage-center cabinets, going for a natural sound with as much level as I can get before feedback. I find that if I am having to do severe cuts with the graphic equalizers, then something else is wrong, because the Meyer cabinets do not require much EQ to sound good."

After checking each downstage mix separately with the sound of his own voice, Spitzig proceeds to bring in the sidefills and try for a balance between the two. "Since I end up having the featured vocal mike coming from five or six different mix sources, and heavily layered in reverb, it is *crucial* to make note of the 'trouble' frequency regions in each mix before the actual show," the engineer continues. "This gives me a good handle on potential feedback frequencies, should any arise during the performance."

Close watch on the program material is kept with an Ivie Model IE-30A real-time analyzer. Mounted on the monitor console, the RTA gives a general display of electrical or acoustical program material, divided into one-third octaves (Figure 10).

A unique switching device developed by David Dansky enables the board operator to access program material from the console's ¼-inch phone jack cue output, or material



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ACOUSTIC MASKING IN STAGE MONITORS — continued...

complicated in practice. The idea is to isolate a zone on stage where the musician will generally reside during the concert, and provide a method by which the sound of his instrument is a bit louder in that zone. If the musician is a bassist, then he has only to worry about equivalent-frequency masking, since there are no instruments that generate significantly lower frequency sounds.

Once a monitor has been selected with sufficient bandwidth (down to about 50 Hz is adequate for virtually any application), then you have only to achieve the appropriate level with the prescribed zone. If you are monitoring a guitarist, vocalist, or other instrument with midrange content, then you must also consider the problem of bass masking. In order to ensure that the monitor speaker will not contribute to bass masking, it should have a slightly elevated midrange output.

The last major problem is creating the isolated zones on stage. These zones are controlled by the polar response of the monitors, and by their physical positioning. Selecting the size of these zones is a balancing act: if the zones are too large, they will interfere with one another and contribute to the masking problem. On the other hand, if the zones are too small, the musician's freedom of movement is sacrificed. The decision about what sort of polar response is acceptable is a judgement call that must be made by the speaker designer.

The new Klipsch KSM-1 monitor has a frequency response of 50 Hz to 15 kHz, ±5 dB, with a midband sensitivity of 102 dB at 1 watt, 1 meter, and maximum continuous output of 125 dB at 1 meter. The average beamwidth is 75 degrees horizontal, and 60 degrees vertical. The cabinet also has a multi-angle configuration that allows more freedom in setting up the stage zones. The frequency response of this system is such that there is a 2 dB increase in sensitivity above 1.2 kHz, a change in sensitivity that virtually eliminates any chance of bass masking from the monitor. The sensitivity specification of 102 dB to overcome the SPL of a PA stack at close ranges, and thereby prevent equivalent-frequency masking while using only a reasonable amount of power.

When selecting a new monitor, three major factors must be considered; frequency response, sensitivity, and beamwidth. The shape of the frequency-response curve should be such that no undue emphasis is placed on the low-end, and a little increase in midband sensitivity is desirable. To monitor bass and synthesizers, response down to about 50 Hz is useful, while response down to 85 or 90 Hz will generally cover the range of most other instruments. Sensitivity should be as high as possible to ensure that an adequate sound pressure level will be achievable. And lastly, a judgement call must be made as to whether the beamwidth is acceptable. Consideration of these factors and an awareness of the mechanics of masking should place the pro-sound contractor squarely in the middle of a well monitored stage. □□□

***Above:* Klipsch KSM-1 on-stage wedge monitor.**

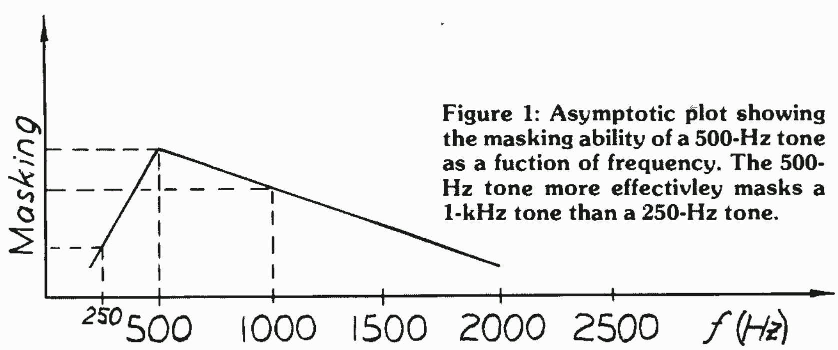


Figure 1: Asymptotic plot showing the masking ability of a 500-Hz tone as a fuction of frequency. The 500-Hz tone more effectivley masks a 1-kHz tone than a 250-Hz tone.

gathered by a calibrated microphone placed out on the performance stage. "A trim pot is necessary to provide equivalent input signals to the RTA when switching between the different sources," notes Dansky. "It is extremely handy to be able to view individual inputs, separate mix output, or the accoustical field all on the same display screen with just the flip of a switch."

Dansky and Spitzig work together as a team to equalize the critical vocal monitor mixes. "It is important to note that the RTA is merely another tool to get the job done," Spitzig advises. "It certainly does *not* replace the human ear. Anyone who doesn't actually *listen* to the effect that his manipulations of the controls has on the mix on stage is doing only half the job." A Meyer UM-1 located at the monitor mix position is used as a cue speaker.

The Show: Opening Night

The first show in a given location is perhaps the most critical one. "This is the time that the orchestra is first sounding out the room acoustics," notes Spitzig. "And we have spent perhaps 20 straight hours doing the load-in, putting up the stage set and sound system, and getting the microphones and cables placed properly.

"Attention to detail is important: cables must be secured, monitor cabinets placed correctly to the inch, and a thorough sound check done with the whole orchestra. *Only perfection counts.* We usually have the local

Figure 7: A Yamaha MS-10 self-powered mini-monitor offers a reinforced piano signal to the conductor.

Figure 8: The monitor position (*stage left*). A Soundcraft 800B console forms the heart of the on-stage monitoring system.

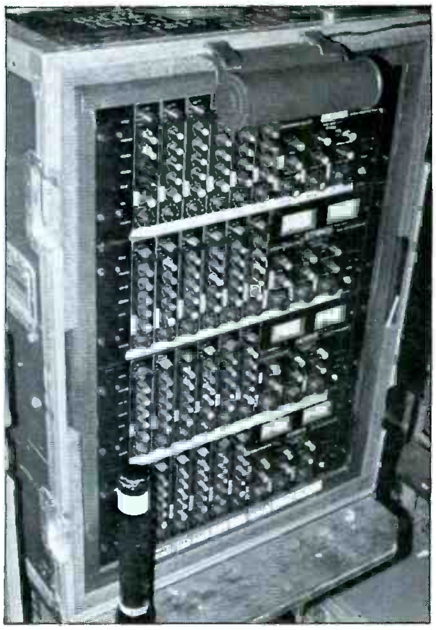


Figure 9: Yamaha M406 mixers serve as submixers for the percussion instruments, trumpets, French horns, trombones, flutes, saxophones, violins, cellos, and background vocal microphones.

press on hand to hear the first performance, and it is a new group of string and horn players to get used to. So the first night at a given showroom is critical."

Out in the house, David Dansky finds that his mix must complement the on-stage sound. "No matter what kind of act you are working with, the casino showrooms are usually small enough that some leakage of the stage sound out into the audience area is unavoidable," he explains. "Being able to build on top of that sound — instead of trying to overpower it with brute force — is the key to successful mixing in this type of situation. Some of the drum sounds, electrical instruments and vocal monitors will be heard in the room. The house mix

Figure 10: An Ivie IE-30A real-time analyzer, mounted at the monitor console position, offers a third-octave readout of program material.

Englebert Customized STAGE MONITORING

must incorporate those sounds, *not* fight them."

Violin, cello, trumpet, reed and trombone submixes from the monitor position are passed out to the house console, along with signal splits of all primary inputs. A 50-pair snake cable, fitted with quick-release AMP connectors, makes setup quick and sure. A mini-connector patchbay at the monitor mix position allows for changes in signal routing, both inputs and outputs.

Two multi-pin connectors have been installed permanently on the back panel of the 800B stage console, to tie the board into the stage input snake system and the auxiliary patchbay rake. Solid-core, gold-flash multipins were used in the construction of the snake system for reliability. An on-board pin matrix provides selectable signal routing for the 800B. Twenty-pair connectors tie the equalizer and effects racks into the master patch panel.

Dansky uses a Yamaha PM-2000-32 console for front-of-house sound. "The PM-2000 is standard in most casino showrooms," he notes. "On this tour, I only had to bring in a rental unit once; most of the venues that we have played to are already equipped with one."

At Harrah's, Dansky had set up a pair of A-1 Audio's RLB (Rear-loaded Bass) cabinets and two radial horns per side to augment the overhead house loudspeaker system. "Stereo imaging can be an important addition to a showroom house mix," the engineer notes. "One trick that I have found helps to really improve the sound of the background vocal group in the house is to pan the submix left and right, while passing one side through a Lexicon DL-3 digital delay

Figure 11: Meyer M-1 and M-3 processors and BGW 750W stereo power amplifiers drive various Meyer on-stage loudspeaker monitor cabinets.

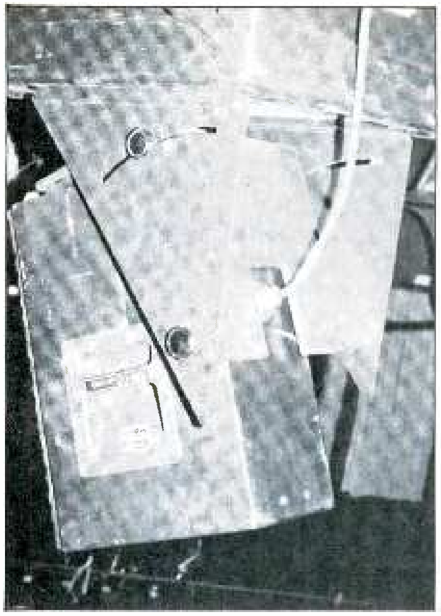


Figure 12: A custom-crafted adjustable cradle, designed by shop fabricator Lou Mennick, was assembled to hold the Meyer UPA-1 cabinets.

unit set at about 12 milliseconds — it helps to give a very natural chorus sound to that part of the mix."

Tape cues, both dialog and special effects, are a major part of the finale to Humperdinck's show. "I have two Revox machines rolling simultaniously, with an A/B selector switch available in case of tape transport failure," explains Dansky. The taped program material is fed up to the stage via the monitor console.

[Refer to the October 1984 issue of *R-e/p* for a full description of "Applications and Utilization of Click Tracks to Provide High-Quality Sound and Synchronization in Live Performance" — *Editor*.]

The Hardware

The A-1 Audio system leased to Humperdinck's show comprises a total of 58 cases, weighing 9,672 pounds. The approximate volume of these road cases is 630 cubic feet. According to a computer-generated manifest, the system's replacement value is $257,000. The manifest allows the road crew to keep close tabs on the system for which they are responsible.

"We do a lot of foreign travel with some of our accounts," notes Dansky. "It is important for the system to be packaged in cases that are small enough to travel easily by air, and to be checked as excess baggage when necessary."

The monitor system includes a total of 11 BGW Model 750B stereo power amplifiers, housed in sturdy road racks with removeable front and back panels (Figure 11). Two BGW Model 100Bs are provided for driving small loads.

The system is equipped with a field repair kit (including phase checker, soldering station, and spare loudspeaker diaphragms) as well as a multitude of spare parts. "Since we are often in one venue for a week at a time, there is *no* excuse for not fixing a piece of gear if it goes down," Dansky offers. "When you take a system out on a string of one-nighters, it can be rough. But we like to be able to keep things going. And, having A-1 offices with equipment in the major casino cities makes it easy on a tour such as this one."

Yamaha Q1027s and UREI Model 539s are supplied for equalization needs. A Sundholm Model 2100 miniature octave EQ unit was available for channel-insert processing on the bass guitar and kick drum. A spare Soundcraft 800B power supply also was on hand.

A-1 supplies a complete power distribution system; tie-in is made at each new venue. "We use whichever of the two phases is cleanest," notes Dansky. "That feeds the minidistro panel, which is equipped with metering to monitor the voltage between neutral and hot, and neutral to ground. Each leg run out of the distribution panel caries both phases, and uses four-pin twist-lock connectors for getting the AC power out to the stage gear and the audio system."

Conclusions

This stage monitoring system was loud and clean. The impeccable attention paid to orderliness and detail was impressive. And the array of custom-tailored devices and technical "fixes" was interesting to see. One such device, a cradle for the Meyer UPA-1A cabinets (Figure 12) was designed and prototyped by A-1's shop fabricator Lou Mannick on less than 24 hours' notice.

Servicing showroom entertainers requires personal service, the right hardware, and each performance must be a "10" when it comes to sound mixing. Two-show nights in a crowded resort hotel can definitely be a challenging environment for a board operator.

Over and above the quality of the hardware, this particular sound crew possessed a desire to "make it right" that was noticeable. Monitor cabinets were constantly being repositioned in quest of the perfect on-stage balance. A re-equalization of each mix was done nearly every afternoon prior to showtime.

"Staying healthy and getting plenty of rest is important," notes Spitzig. "This is not a 'set it and leave it gig.' Staying on top of the gear, and the show, all the time, is what makes it all work." ■■■

SYSTEM EVOLUTION:

Monitor Changes for Engelbert Humperdinck's 1985 Tour

For 1985, A-1 Audio has made several changes in the stage monitoring system supplied to Engelbert Humperdinck. "Having a system on the road doing only one-nighters can be tough," states house mixer David Dansky. "However, when you take a show into a particular venue for a week at a time, it is possible to really fine-tune everything, and make improvements in the show's sound as you go. You don't have to wait until everything gets back into the shop at the end of six months, or whenever. In that sense, taking out a system with this kind of show is like having a concert-sound laboratory — you keep experimenting until it's right."

Such experimentation during the 1984 tour has led to changes in the sidefill monitor speaker system: more cabinets and amplification now offer greater headroom. A new type of Countryman Isomax II miniature condenser microphone has cleaned up the stage look considerably, and improved the sound of the orchestral and rhythm sections. A rolling electronics rack offers improved speaker system equalization capabilities. And (you guessed it), a different personality has been brought in to handle the complex monitor system.

Expanded Sidefills

The single Meyer MSL-3 cabinet on each side of the stage has been doubled, with Meyer USW-1 subwoofers added for extra low-end presence.

"What Engelbert really wants is a *house* sound on-stage," explains Dansky. "In trying to give him that before, the system was being pushed sometimes right to its limit. By doubling the sidefills and adding subwoofers, the system has more head-room. It doesn't work as hard to give him the presence he wants. The amplifiers don't clip, and the sound is cleaner. And the overall stage sound is now somewhat lower than before, believe it or not!"

Monitor mix engineer Ken Newman previously has worked Humperdinck's show from the house-mix position, and doubled as production manager as well in the past. "The monitors really are the key to the sound here," he confides. "A really up-front basic rhythm mix featuring piano and percussion gives him what he needs to sing to. The subwoofers are available for added presence on the more active numbers."

Microphone Array

The conspicuous forest of microphone stands usually seen on concert stages has been replaced by a microphone product that is nearly invisible to the audience. Developed by Carl Countryman, the unique new Isomax II hypercardioid condenser microphones are equipped with mounting brackets that in many instances, do away with traditional stands; 37 of these mikes are in use with this show to cover the horn and string sections, the percussionist's

For the latest-generation of stage monitor setup for Engelbert Humperdinck, each side of the stage is flanked by a pair of Meyer MSL-3 enclosures, with a USW-1 subwoofer mounted on top (*pictured left*). The stereo mix for these cabinets is intended for the featured vocalist, and serves only the performance area. The drum set (as well as string, horn and percussion sections) has now been provided with miniature Countryman hypercardioid condenser microphones (*pictured above, and inset detail*).

setup, and all of the drum set except kick and snare.

"These mikes are easy to tuck into place anywhere, and they are visually pleasing," Dansky notes. "However, the greatest advantage I have found with this system is the uniform frequency response of the microphones. Instead of having a dozen different types of mikes up there for different applications, I just have one. It makes the sound of the band and orchestra much more consistent."

A Nady Model 700 VHF wireless microphone, equipped with a Shure SM-87 capsule, has been added to the show for use during a featured dance segment.

Rolling Equalization Rack

A novel approach has been taken to solve the problems associated with the critical "tuning" of the various monitor mixes. Newman has equipped his wheeled monitor EQ rack with extra-long cables, allowing him to position the rack center-stage when setting the various speaker zone mixes. A 50-foot, 20- and eight-pair snake cable setup enables him to roll the rack to any point on the downstage area.

"It might seem like a strange idea, but it is quite helpful," the engineer explains. "Instead of talking into the mike, then having to give some instructions to another person standing over at the monitor board, I can get *exactly* the sound I want without having to 'translate' my requests to somebody else. I guess the next step in this direction would be a digital control head for the EQ rack, something like a Lexicon LARC device."

Sound of the Show

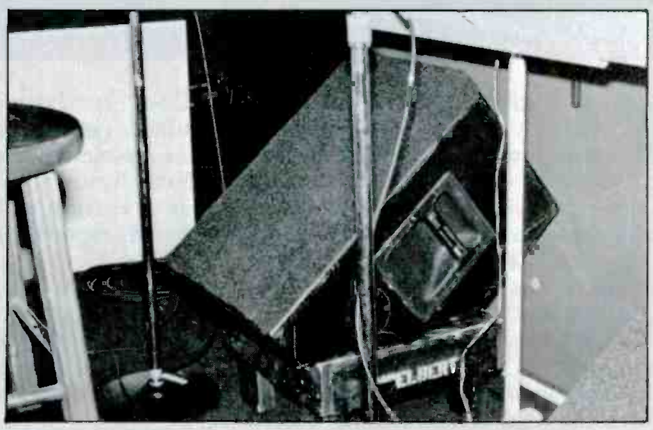
Having heard this same show in late 1984, I was quite curious to hear what results the expanded stage monitoring system would have on the house sound. I purposefully positioned myself at a table for the dinner show that seemed to be equidistant from the installed house-sound system, the temporary house-fill stacks, and the downstage monitor line.

I noted with interest that, despite the increased size of the stage-speaker system, there did seem to be less interference with the house mix. The sound of the monitor system appeared to be confined more to the on-stage area than before, and what sound did escape into the seating area seemed to have a intelligibility.

"This show is easier to mix now," Dansky offers. "Microphone uniformity on so many of the instruments leaves less room for error, and gives me a more even sound. However, it is almost a paradox that enlarging the sidefill monitor system has led to a *lower* overall stage volume. It is curious, but it works."

Today's stage monitoring systems are far more complex than most main house speaker systems of only a decade ago. And they offer better audio quality, due to improved transducers and specialized signal processing equipment. As advanced systems such as this one are brought into venues that offer an installed system, stage technicians and audience alike cannot help but notice a difference between the bold system and the newer one.

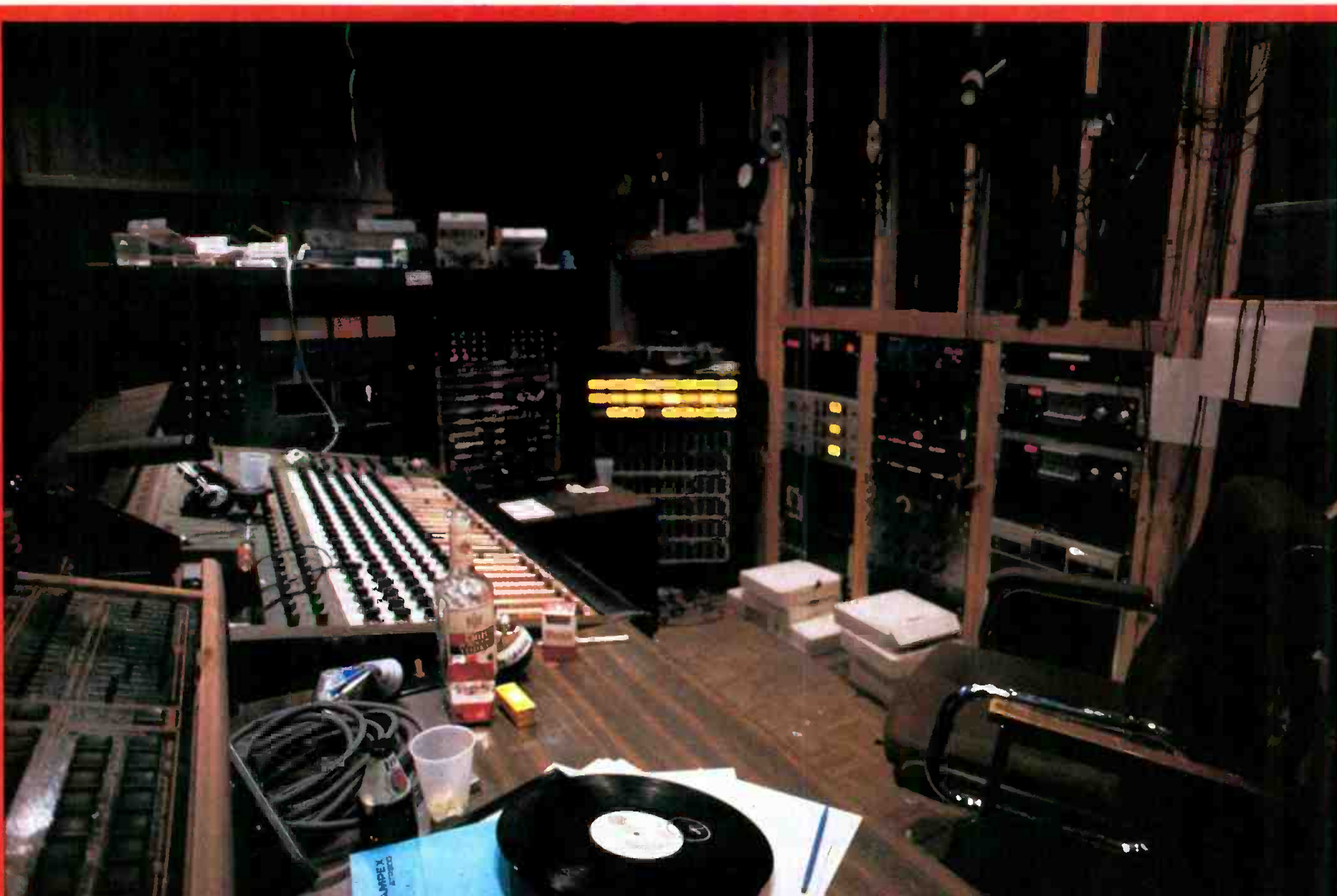
"What many people don't realize is that an entertainer has heard it all by time they reach the point in their career of playing the casino showrooms," states Dansky. "Any show that can do a week's stand in Vegas or Atlantic City or Tahoe has been around long enough to have heard just about every type of system there is. They expect the best, and they can pay for it. They know what they want to hear. Trying to help those entertainers hear what they want is what makes this job fun." □□□



Various types of floor slants are currently being used to provide on-stage monitoring for Engelbert Humberdinck, including (*clockwise from top-left*): Community Light & Sound NC-12s; JBL Model 4602s; Meyer Sound UM-1 UltraMonitors; and Yamaha S2115Hs.

EDWARD VAN HALEN'S 5150 STUDIO

by Howard Weiss



Photography by Bonnie Parkinson

The Eighties have seen proliferation of personal-use recording studios tailored to the requirements of leading artists and record producers. It is important to understand some of the basic reasons for the increased popularity of home studios in the production of professional product:

- An abundance of used state-of-the-art equipment at a reasonable price;
- A trend towards the use of more electronic musical instruments, and a related decline in the use of acoustic instrumentation;
- With overdubs consuming 50% or more of a product's budget, it is cost effective for a project to be taken to a personal-use studio (suitably equipped);
- User creativity is increased when projects can be done without studio-time restrictions; and
- New digital reverb technology enables a variety of room sounds to be created, regardless of the actual studio size.

The home studio, with proper layout and execution, is able to handle an increased variety of projects, yet still capable of producing a quality product.

While the creation of a quality personal-use studio shares similar problems to a commercial facility — construction techniques, isolation, room geometry, etc. — unique problems are frequently encountered. (One of which is the studio's ability to contain sound pressure levels in excess of 130 dB without disrupting adjacent neighbors.)

— the Author —

Other studio-design projects **Howard Weiss** has been involved with include Cherokee Studio 3, Los Angeles, 335 (Larry Carlton's personal-use studio); Sunset Sound Studio 1; Lion Share Studio B; and Hog Manor (David Paich's home studio).

Personal-Use Criteria

One of the first steps in pre-planning the facility is to list the requirements and clarify the goals the user expects from the studio:

- Is the facility to be used just for overdubs and mixing?
- Do keyboards represent a major usage of the studio? If so, does sufficient space need to be provided in the

The Professional Cassette Decks

Tandberg's new Series TCD 900 is a superior and cost-efficient alternative to the (unprofessional) practice of using inferior home tape decks for Professional applications. These new Professional Cassette Decks offer unparalleled sound capability, advanced mechanical and electronic design, plus extraordinary control flexibility based on an 8-Bit microprocessor with 32K of EPROM memory. These are the type of quality products for which Tandberg is well known, and are designed and built in keeping with the company's more than 50-year reputation for quality, performance and long-term owner loyalty.

TCD 910
Master Cassette Recorder

TCD 910 is designed to replace both reel-to-reel and cartridge machines in many applications, and is capable of producing tapes at sound and silence levels beyond that required by broadcast and studio requirements. Combined with its extremely accurate real time counter and sophisticated autolocator functions, this machine is truly a multipurpose cassette recorder.

Features include:

- High precision, rugged 4 motor tape transport with direct load, instant access cassette positioning.
- Discrete three head system with built-in record azimuth adjustment is combined with Tandberg's proprietory Active Phase Correction Circuitry, exclusive discrete, wide band electronics, plus the highly regarded Actilinear II and Dynec systems. In addition, the latest generation Dolby B and C noise reduction processors are utilized.
- All audio circuitry uses high spec polypropylene capacitors and metal film resistors.
- Built-in autolocator with 10 cue points in real time, auto cut search and cue/review.
- Auto stop and/or rewind after cut.
- Electronically balanced XLR input/output connectors.
- Front panel bias and record current adjustment, with built-in oscillators.
- Optional RS 232 computer interface, infrared wireless and hard wire remote with fader start.
- Wide range of options and accessories.

TCD 911
Cassette Playback Deck

The TCD 911 offers the same quality of construction and design as the TCD 910. Its special features include:

- Playback pitch control.
- External playback azimuth control, for optimum performance from *any* pre-recorded cassette.

In a multi-deck studio situation, the TCD 910, combined with the TCD 911, makes for the ideal match of performance, reliability and price.

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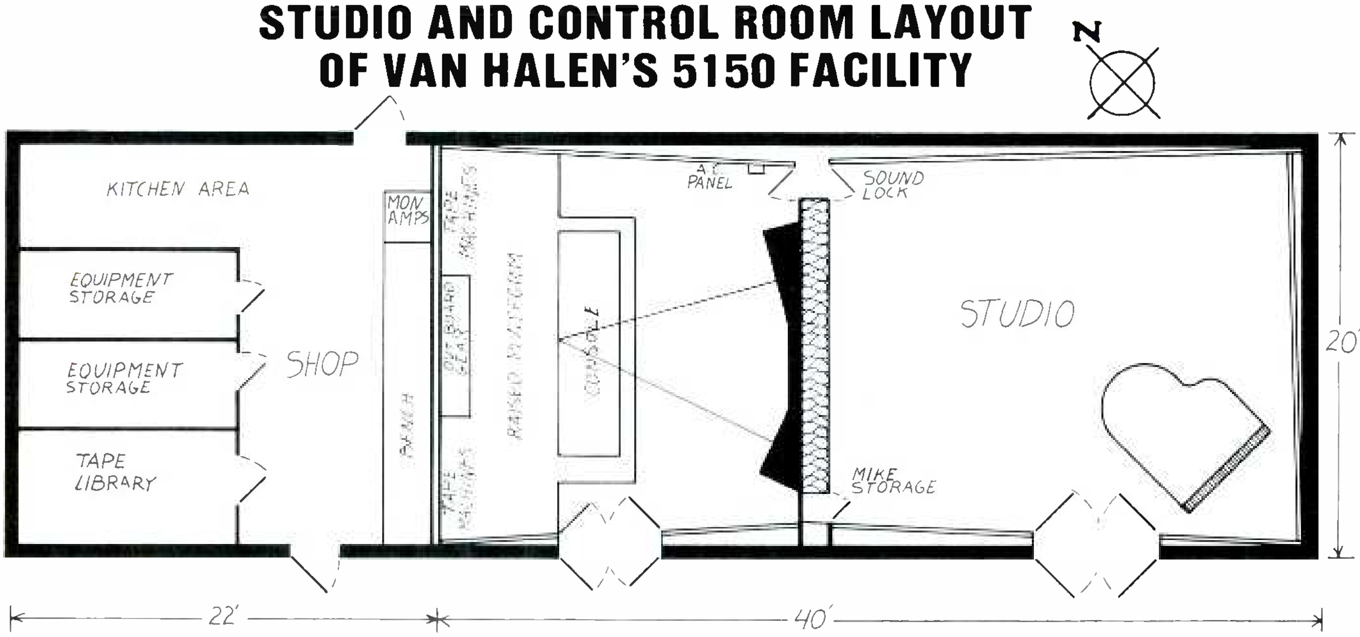
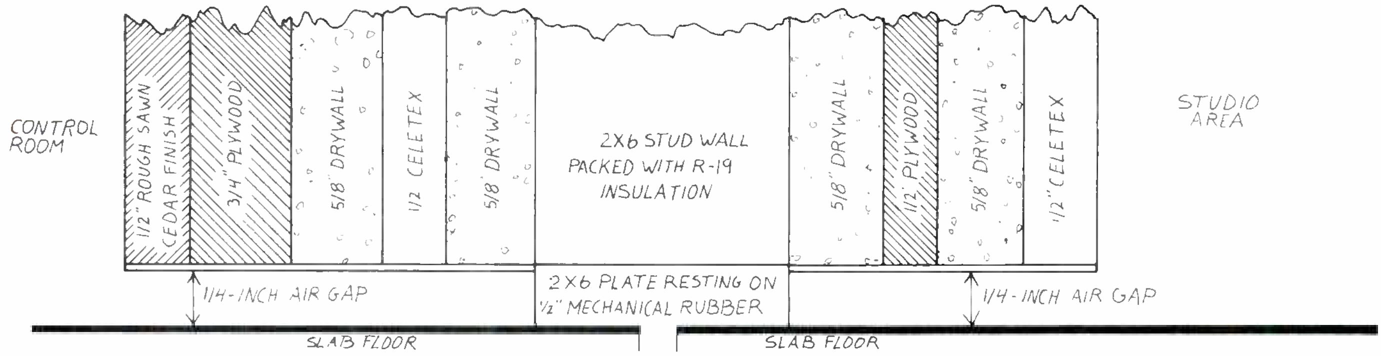
For additional information circle #35

STUDIO AND CONTROL ROOM LAYOUT OF VAN HALEN'S 5150 FACILITY



CROSS SECTION OF STUDIO/CONTROL ROOM WALL TREATMENT



STUDIO DESIGN AND CONSTRUCTION

control room to accommodate them?
- Do they need a 24-track studio?
- Budgetary considerations; and
- Future expansion requirements.

One begins by finding a suitable location for the room. Size requirements and sound containment techniques will dictate construction practices. At this point, I can't overemphasize the need to consult a competent architect and studio construction specialist familiar with all aspects of studio construction techniques.

A good case in point was the construction of 5150 (Police code for "Mental Case"), the studio we built recently for Edward Van Halen, and the recording and mixing venue for his *1984* album. Design requirements for the studio were set down by Ed and his engineer, Donn Landee. Basically, Ed wanted a room where "we could make records;" a place he could use at any hour and not be concerned with disturbing anyone. Donn wanted "a semi-professional 16/24 track studio to accommodate tracking, overdubbing, and mixing." The control room was expected to sound correct without the use of monitor equalization or traps, while built-in control room monitors were chosen because of the high listening levels required.

STUDIO EQUIPMENT LIST

UREI 24/12 console
3M M56 16-track
Ampex MM-1200 24-track
Ampex ATR-800 two-track
Ampex ATR-100 two-track
JVC 8200U U-Matic video recorder
Two Studer A-710 cassette recorders
Revox B-225 CD player
JBL/Augspurger monitor system
H&H power amplifiers
EMT 140ST plate reverb
Quantec Room Simulator
Four UREI 1176 limiters
Two Teletronix LA-2A limiters
Eight Valley People Kepex gates
Lexicon Super Prime Time
MXR delay Time
Lexicon Prime Time
Two Eventide Harmonizers
Two Lang PEQ-1 equalizers
Two Pultec MEQ-5 midrange equalizers
Two UREI 550 filters
Two Neumann U-48s, AKG C-12, four Neumann KM-84s, two Sony C-37As, two Neumann U-87s, two Sony ECM-50s, AKG 414, eight Shure SM-56s, four Sennheiser MD421s, and two Sennheiser MD441 microphones.
1912 Hamburg Steinway "B" piano (MIDI-equipped) ☐☐☐

Good sound isolation between control room and studio also was mandatory.

Technology was not the main factor in choosing equipment; the choices were based on what sounded good and what was available at local parts stores.

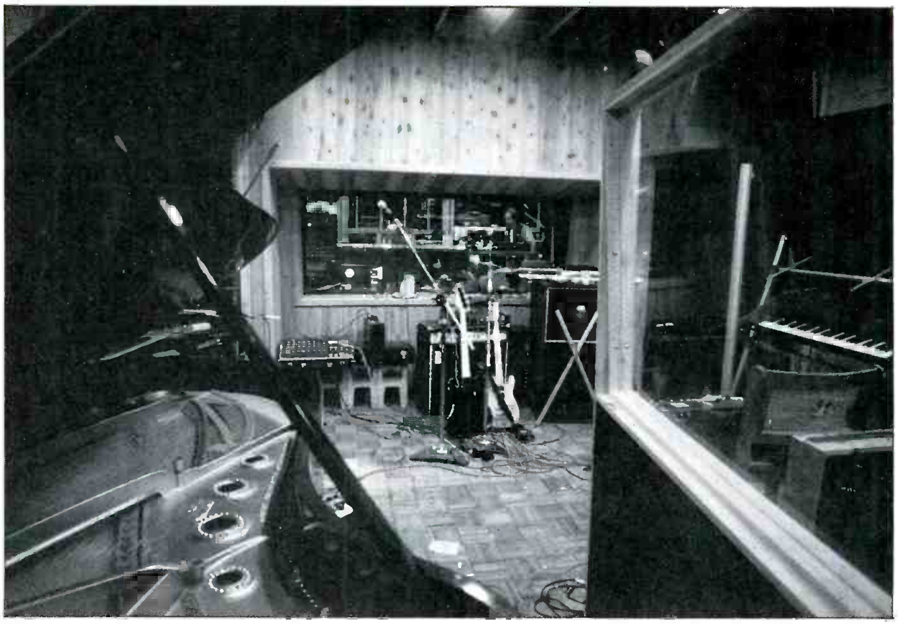
We incorporated the talents of Ken Deane, representing the Mt. Baldy Lodge, and Frank Latouf, aka "Guido," a studio specialist. The efforts of Drew Bertinelli and Ron Fry were combined in the basic construction. Kaplan Electric was chosen for the electrical installation, and Carlos for the air conditioning installation.

The Van Halen home, surrounded by tall trees and barbed wire, is located in the hills above the San Fernando Valley, north of Los Angeles. At one end of their property existed a guest house with an unused room. Initially, we considered this as a possible control room site. After some investigation, however, we decided to gut the structure and build the studio from the ground up. Consequently, a 20-foot wide by 40-foot long structure with a shell ceiling height of 18 feet was constructed. Based on isolation requirements, 10-inch re-inforced block construction was employed, all block cells being pumped full of concrete. A five-inch thick re-inforced slab also was poured.

While construction proceeded, we began the task of modifying the console and 3M 16-track. The UREI console had been in service at United Studio A in Los Angeles for many years. Although it had great sound, for our needs it required many modifications and additions. The decision was made to rewire the entire console, and incorporate the following modifications: a new 520-position patch bay; a new monitor selector panel; new stereo and cue busses utilizing Jensen 990 amplifiers; eight additional line inputs; and phantom power. (Extra parts, such as patch bays and vacuum cleaners, were deposited in the septic tank for semi-permanent storage.)

The deadline we faced required us to work at all hours of the day and night. Valerie Van Halen suggested that "Club Daiquiri's" were to be consumed in great quantities to maintain the proper state of mind, Ken and Donn insisted on the continuous viewing of *Blazing Saddles* while consuming daiquiris.

After the basic shell was completed, the interior construction began. Possibly it was the daiquiris, but Ed wanted the control room to be sited on the north side of the structure. "I want to face North while playing." That was the end of that.



The recording area at 5150 measures approximately 17 by 23 feet, with a ceiling height that slopes from 12 feet at one end to 10 feet at the other. A wide range of studio instruments is available for visiting musicians, including a vintage 1912 Hamburg Steinway B grand piano, pictured right.

The concrete slab layed between the control room and studio was cut in three places to decouple low-frequency transmission between the two areas. In addition to conventional practices for studio construction, we used our intuitive judgement in room geometry and surface treatment, a factor that is sometimes over-shadowed in current design philosophies. The present trend toward utilizing computer projections in predicting room preformance has a major flaw: Humans use The Room, *not* Computers. Experience in a multitude of major facilities has enabled us to judge with a good degree of accuracy the dimensions and treatment that will best suit a particular room. There are numerous situations where the computer cannot factor the parameters that yield optimum room performance.

Interior Construction Common to Control Room and Studio

The space was divided into two areas, a control room being built 17-foot wide by 14-foot deep, with a ceiling height sloping from 12 to 10 feet. The remainder became the studio, measuring 17-foot wide by 23-foot deep sloping again from 12 to 10 feet. The slab was covered with half-inch particle board and half-inch parquet flooring. The interior walls are constructed from 2×4 studs on 16-inch centers, liberally blocked and insulated with R-19 fiberglass insulation. Wall plates rest on half-inch mechanical rubber. Stud walls were covered with ¾-inch plywood, ⅝-inch drywall, half-inch Celetex, and ⅝-inch drywall. All seams were staggered, each layer glued, and the drywall taped.

The roof consists of composition

. . . continued overleaf —

material on 1⅛-inch plywood over 2×12 joists resting on the exterior walls. Roof joists were insulated and the bottom covered with ⅝-inch drywall. Five feet below this a secondary ceiling was constructed on 2×12 joists, insulated with R-38 fiberglass insulation, attached to the exterior walls and isolated with half-inch mechanical rubber. Both surfaces are covered with two layers of ⅝-inch drywall and ¾-inch plywood. The five-foot space between this secondary ceiling and the roof is used for air conditioning ducts, electrical runs, and a home for our Green EMT. A false 2×6 ceiling was constructed at a height ranging from 10 to 12 feet to accommodate lighting and air conditioning.

Special Construction

Two feet from the rear of the control room, yet another wall was constructed to house three, 48-inch high racks of outboard gear, with tape machine soffits on either side. This wall was then carefully tested and found to be bullet-proof (.44 magnum JHP at two feet). Left-to-right acoustic symmetry was maintained throughout the construction of the control room.

The control room platform was built using 2×6 joists on 16-inch centers, resting on half-inch mechanical rubber; the joists were packed with R-19 fiberglass insulation. The subflooring consists of a layer of ¾-inch plywood, half-inch plywood, ¾-inch particle board, and half-inch oak parquet flooring.

The control room/studio wall is actually two isolated walls, each consisting of 2×6 studs on 12-inch centers, and insulated with R-19 insulation. The walls are heavily blocked, resting on half-inch mechanical rubber with six-inches of air space between them. The sides facing the air space are covered with half-inch Celetex, ⅝-inch drywall, half-inch plywood and ⅝-inch drywall. The sides facing the rooms are covered with ¾-inch plywood, ⅝-inch drywall, half-inch Celetex, and ⅝-inch drywall.

Two 4×12 beams span the length of the control room studio wall — in addition to picking up the bearing load of the false ceiling, the beams support the monitor cabinets. Two, 3- by 8-foot panes of glass were used in the control room window. Between them, a Piranha smoking a Camel, and a Club Daiquiri were installed. On the studio side, a ¾-inch pane of glass was used with mechanical rubber stops, while on the control room side a ⅝-inch pane of glass was used.

Soundlock doors used on either side of the control room to studio entrance were constructed using 1¾-inch solid-core door, with the soundlock side glued to ¾-inch particle board, and the other side faced with ¾-inch rough sawn cedar. Bottoms of all doors are equipped with Pemco automatic door bottoms. Interior doors are solid-core two-inch finished with rough-sawn cedar; exterior doors are two solid-core two-inch, laminated using mechanical rubber.

The monitor cabinets were roughly set in position above the window. A string line on-axis with the midrange horn was used to establish precise convergence at the mixer's position. (This also corresponded to side-to-side center at the console bolster.) A 4×6 beam was sledged between monitor cabinets and ceiling joists to further couple the monitors to the room. Once the monitor angles were set, a three-inch layer of pour-stone cement was added under each cabinet. The pour-stone's function was two-fold: to provide for the monitor cabinets at a complex angle; and to enhance low-frequency coupling of the monitors.

The Augspurger cabinets occupy 16 cubic feet, and each contain two JBL Model LE-15 low frequency drivers, toeing into each other at a 30-degree angle; angling the woofers in this manner tends to smooth out response in the 70 to 150 Hz region. JBL Model 2441 (375) midrange drivers with Model 2390 horn/lens assembly also were used, along with Model 2405 "Super-tweeters." The system incorporates a low-level, 18 octave White passive crossover operating at 800

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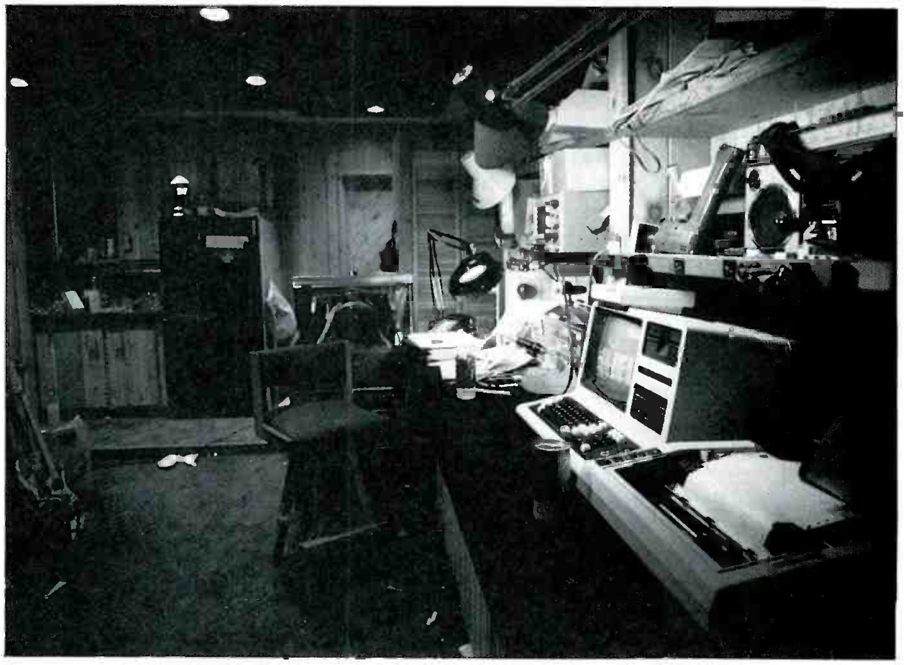

The maintenance/repair shop and kitchen area at 5150.

Hz. A high level, 6 dB per octave passive dB per passive R/C network crosses to the 2405 at 7 kHz. An H&H V-800 is used for low-frequency amplification, while a V-500 powers the midrange and high-frequency units; #10 stranded copper wire was used for all speaker runs. After various pink-noise tests, no room EQ was found to be necessary. Provisions for small speakers are built-in, but they have never been used.

Grounding Considerations

Our studio grounding system consists of two, eight-foot copper ground rods. The installation procedure is as follows: a hole is excavated to a depth of four feet; the ground rods are driven into the hole leaving one foot exposed above ground level; 10 pounds of copper sulfate crystals are then mixed into the excavated soil, and repacked around the ground rods. A 50-foot, #00 stranded copper wire then ties this ground plane to an isolated ground distribution bus in the electrical panel.

A #2 stranded copper wire connects the console ground bus to the isolated ground bus in the electrical panel. Our audio grounding scheme used the console patch bay as the ground reference — lines running to and from auxiliary equipment and tape machines are grounded at the console, and lifted at the equipment. All outboard gear receives its ground from the AC outlet; equipment which was supplied two-pin was modified to three-pin configuration. All patch cords are grounded at one end only.

One unique problem confronting us was KMPC, a 50,000 watt AM radio station sited just three miles from 5150. It was also a necessity that Ed be able to play in any location facing any direction, without hum, RF, or noise problems. Our solution was to build a chicken coop (grounded, of course): standard chicken wire was used in the walls, flooring, and ceiling to surround the entire recording area. The wire enclosure was tied to our central ground, and at no point touched any electrical conduits. The concept worked flawlessly.

The electrical panel is located between the control room and studio, away from tape machines and other equipment that is field sensitive. 5150 uses a 220V-100A panel with ample circuits for future expansion. We ran three #00 stranded copper wires from the main house to 5150 (a run of over 200 feet). The use of isolation transformers was avoided because such devices add harmonic distortion to the AC line, causing power supplies to run hotter. Any isolation transformer also raises the impedance interconnect between studio and utility power, thereby increasing the possibility of RF interference. Extra care was taken in balancing the load of the 220V line.

All power outlets are "orange hospital grade" type, where the ground pin is isolated from the box. A #12 green wire runs from each outlet back to the isolated ground bus in the electrical panel. Also, separate neutrals were run for each circuit. The only contact between conduits is at the electrical panel. The lighting system consists of eight auto-transformers

5150 studio owners Edward and Valerie Van Halen

(800-watt Luxtrols): four each for the control room and studio.

Air Conditioning and Heating

Air conditioning was accomplished by use of a modest sized, two-ton shock-mounted roof unit for both heating and cooling. A custom sheet-metal system utilizing 24- by 24-inch supply and 12- by 24-inch return ducts was built. To eliminate acoustical transmission between the rooms, sound traps were fabricated and installed in the supply and return ducts to the control room and studio. All supplies and returns were decoupled from the rooms using flex-ducts. Ducting was insulation lined and isolated from structural contact.

Due to the high "R" value of the interior walls, and the high heat retention of the exterior walls, a 64-degree inside temperature is maintained while outdoor temperatures may vary anywhere between 29 and 110 degrees Fahrenheit. A petite air conditioning unit is all that is required to remove heat generated within the room.

By this time, we were anxious to hear something in the room. All we had was a cassette copy of "My Mother is a Space Cadet." (Ed and Donn had just produced this record for Dweezil Zappa.) We were blown away with the sound of the room, even before the surface treatment was finished.

Finish Treatment

The control room was finished with rough-sawn cedar on the front wall and both side walls. Burlap-covered Tectum was placed on the walls from the console bolster to the rear of the control room. The rear wall was left with studs exposed and covered with Airflex insulation. The ceiling was filled with R-38 and covered with black burlap.

The studio finish consisted of rough-sawn cedar on the control room/studio wall, exposed R-38 insulation in the ceiling, and sparse covering of the remaining walls with Sonex acoustic foam panels and Airiflex insulation. Our attitude on the finish treatment was to stop the finishing work when things sounded right. There is very little finish.

In addition, Ed and Valerie's two-car garage was commandeered to become a shop, tape library, instrument storage, and kitchen. The adjacent guest house also was taken to become the lounge.

The Acid Test

As we were comfortably sipping B-52s at 64 degrees — a powerful Halen concoction made from equal parts of Kahlua, Baileys and Grand Marniers — the first sessions in the studio resulted in the creation of the song "Jump" and the album *1984*; all subsequent recording and mixing of this album was done at 5150.

Other 5150 projects include the production of Van Halen's portion of the 1983 *US Festival* radio broadcast; *StereoSeptic*, a recording made when the unwanted studio hardware found a watery grave in the septic tank; the score for the CBS Movie of the Week, *The Seduction of Gina*, the score for the Universal motion picture *The Wildlife;* and the Grammy-nominated recording of "Donut City" from *The Wildlife* soundtrack album. The success of these projects reflects the talent of all involved.

Let's leave the last word to Ken Deane: "This is a functional studio; it isn't designed to be in *Better Homes & Gardens.*" ■■■

RECORDING AND PRODUCTION TECHNIQUES FOR DIGITAL SYNTHESIZERS

A Profile of Denny Jaeger's new Personal-Use Facility for Commercials and Soundtrack Production

by Adrian Zarin

Nestled on a moutain side in a rustic San Francisco suburb, Denny Jaeger's secluded home is the last place you'd expect to find a busy film and television facility. Although it occupies no more space than the average family's basement rec room, the composer's one-man digital synthesis facility has produced the music for a wide range film projects, including *The Hunger* and *What Waits Below*; television shows like *Capitol* and *The Powers of Matthew Star;* plus countless jingles, most notable being Levi Jeans' animated *Movin' On* series.

As a long-time consultant to New England Digital Corporation, manufacturers of the Synclavier range of digital synthesizers, Jaeger has built his facility around a fully-configured Synclavier system. In this respect, his studio is somewhat typical of the many synthesizer-based scoring facilities springing up across the country. As digital synthesis developed and became more sophisticated, it has taken over many of the jobs that traditionally belonged to other parts of the recording process. It has become more than just another sound source — another instrument; and the challenge to studio designers has been to adapt the traditional recording environment accordingly.

The extended capabilities of digital synthesis are well represented by Jaeger's own Synclavier system. Apart from its sound-synthesis abilities, the computer-based system incorporates a 32-track digital-memory recorder, plus full sampling and resynthesis facilities. All of which enables the composer to assemble and mix a 32-track composition before he puts one note to tape.

Via sampling, acoustic instruments and other "real-life" sounds can be captured and manipulated by the computer. At the time of writing, the NED Synclavier is only equipped with monophonic or single-voice sampling capabilities; from April, however, it will be able to play back sampled sounds in full polyphony with multiple voices. Equally as dramatic is the system's potential for resynthesis, in which a real-life sound is subjected to detailed computer analysis and then recreated in minute detail by means of conventional digital synthesis techniques.

The ramifications of such digital-synthesis capabilities extend to every aspect of studio design. For one, they completed the transition — initiated by the advent of analog synths — from the tracking room to the control room as a focal point of music-making activity. With the ability to emulate actual orchestral timbres, the one-man composer/producer/engineer can stretch his scope beyond "electronic music," and take on projects in all musical styles.

At the same time, digital synthesis assigns a more passive role to tape machines and consoles, which now mainly serve to capture a composition that has already been fully realized and mixed within the digital synthesizer itself. Also, the properties of digital sounds created by this type of synthesis impose their own special requirements on monitor loudspeakers, consoles and tape machines. All of these considerations emerge in the design of Jaeger's studio.

Studio Design and Outfitting

The facility consists of a modest-sized tracking room, an iso booth, a control room, a small room that houses the studio's computers, and a tape/ equipment storage room. In building the studio into the lower level of his home, Jaeger's goal was to fit it into the existing structure of the house; he was especially interested in avoiding odd room shapes that would diminish the building's resale value.

Working within these restrictions, one of the biggest problems he faced was the acoustic difficulties of having all parallel surfaces in his tracking

and control rooms. In the tracking room, the problem was solved by installing triangular redwood louvers that cover all four walls. The room is also outfitted with drapes made of absorbent cloth, which can be drawn around all four walls. On one of the walls, the acoustic surfacing can be removed to reveal a window that affords a view of the surrounding countryside; the control room has similar provisions for contact with the outside world.

Although they may seem a trivial design detail, Jaeger finds the windows fairly essential to his work. "One of the biggest difficulties of a one-man operation is the tendency to go stir crazy," he laughs. "I'm down here working by myself for hours and hours. So whenever room acoustics aren't vital to the task I'm performing, I'll open up the windows. Psychologically, it seems to help quite a bit."

The floor of the tracking room is divided between two surfaces: hard teak and plush carpeting. An acoustic drape can be drawn between the two areas as well, cutting the tracking room in half; the teak-floor surface can also be covered with carpeting. Jaeger finds the room more than adequate for the occasional vocal and instrumental overdubs, and for sampling live acoustic instruments and sounds into his digital synthesizer.

Jaeger records samples both digitally and on analog tape (30 ips with Dolby) before encoding them into the Synclavier. While sounds can be encoded directly into the Synclavier, he prefers to put them on tape first, for several reasons. Because direct sampling uses up quite a bit of computer memory, the process typically involves juggling several different storage media (as discussed below). It is therefore neither cost-effective nor productive, according to Jaeger, to have live musicians standing by while he performs the necessary computer programming. Also, by recording on tape first, he ends up with a choice of analog or digital samples to be encoded into the Synclavier. This method also enables him to make last-minute alterations in pitch, EQ and other parameters before loading sounds into the Synclavier.

The studio, also equipped with a small, deadened isolation booth, proves useful for recording a particular type of sample. "The iso booth is good for samples where you want absolutely *no* extraneous room sounds," Jaeger says. "I use it for glass and percussion — especially delicate percussion samples such as breaking glass, or tinkling finger cymbals."

Control-Room Equipment

Located just off the control room, Jaeger's computer room forms the heart of the studio, and houses his Synclavier computer and data-storage system. The latter consists of a Winchester disk drive and a Kennedy tape drive, both enclosed in hermetically sealed cabinets. The 20-megabyte Winchester system handles all of the operating software for the Synclavier, and for storing sampled sounds, while the Kennedy drive serves as a backup for the Winchester. Sampled sounds are off-loaded from the Winchester to the Kennedy, and loaded back as

Digital synthesist and producer Denny Jaeger.

needed. Both of these storage systems support the two, 5¼-inch floppy-disk drives located in the control room with Jaeger's Synclavier keyboard and monitor. The floppy drives can store 64 synthesized sounds, up to 30,000 notes of sequencer data, or a short (two-second) digital sample.

Along with the Synclavier computer and drives, the computer room also houses Jaeger's Control Video synchronizer mainframe, Crest 4000 power amps, and Meyer room equalizer (all of which will be discussed in greater detail below). With this much sensitive equipment on hand, Jaeger has taken special pains to make the computer room a stable environment. For example, the room itself is hermetically sealed, constant temperature and humidity levels being maintained by a fanless, freon-tube air-conditioning system. Here, as throughout the studio, isolation transformers

Jaeger's New England Digital Synclavier synthesizer, video-display unit and keyboard controller are housed in the main control room (*left*), while a companion Computer Room houses the main computer processor frame, a 20 Mbyte Winchester hard disk drive, and a Kennedy Tape drive. The latter units are both enclosed in hermatically-sealed cabinets.

All photography ©1985 by John Sievert

SYNTHESIZERS IN THE STUDIO

guard aginst power surges.

In designing his control room, Jaeger again had to sidestep parallel surface problems in order to arrive at an objective listening environment for his projects. "My basic feeling was that I wanted an efficient, good-sounding room without having to spend half a million dollars on the structure," he offers. "Symmetry is very important in a room, and I've tried to preserve symmetry here. Flutter echos [the rapid bouncing back and forth of HF sound between parallel surfaces] have been totally removed; and so have most other high-frequency echoes.

"Low frequencies are more difficult to control because the length of a low-frequency cycle could be as much as 40 feet — which is longer than this whole room. One of the things that helps me here is that his control room is 60 feet up in the air. [The house is built into a mountainside on pylons.] The space below the floor itself therefore absorbs some of the troublesome low frequencies."

Because the control room has a bi-level floor, and the console — located in the center of the room — is itself slanted, the room has few parallel surfaces on the perpendicular plane, Jaeger explains. "The area behind the console is the only one where there could be a parallel-surface problem. We analyzed it and found that there wasn't anything going on in this area that was affecting frequencies in a critical way. The low-frequency standing waves that are in this room are so low that they're not a concern — unless you are located all the way against the back wall. When I'm mixing, though, I'm never positioned seven feet behind the board, up against the back wall, so this is never a problem."

Both side walls of the control room are fitted with multiple panels of Owens-Corning 705 absorptive insulation arranged in different, non-parallel patterns on each wall. The entire back wall is covered with a single sheet of Owens-Corning 705, while the front wall is surfaced in wood paneling. It is broken up by a large, angled glass window looking out onto the control room, and by a large shelf located just above the control room window and running the entire length of the front wall. The shelf holds the studio's principle monitor loudspeakers, a Meyer Sound ACD system.

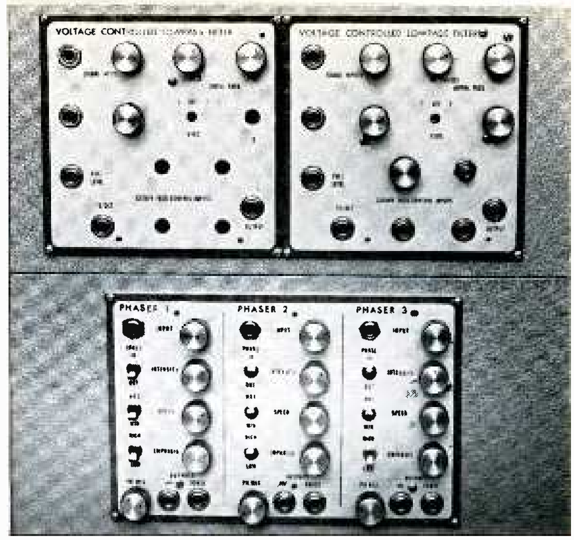
Jaeger's choice of monitors was motivitated by acoustic conditions in the control room itself, and by the nature of his work in digital synthesis. "The Meyers are very transparent room monitors," he notes; "and I felt I needed that transparency. I find that this system — being a two-way system — is cleaner, because you only have one crossover point. The very nature of digital synthesis enables me to build up *very* complex tracks. I wanted speakers that wouldn't color any of the tracks, and that would let me hear each individual part in a mix."

The Meyer system also plays a role in shaping the acoustic environment of the control room. A prototype CP-10 Complimentary Phase parametric room equalizer was installed at the studio, and used by John Meyer to tune Jaeger's control room around the speakers' performance.

"The EQ can be used to correct whatever the structure of the room itself hasn't been able to deal with," explains Jaeger. "I told John that there were some inequities in the room and that some things were not in perfect symmetry; and he replied; 'That should be no problem. We'll just equalize for that.'

"Meyer feels that you can't really tune a room with a graphic EQ, although this is traditionally what has been used. Though I'm not an expert in acoustic design, I'm inclined to agree. The frequencies that have to be adjusted in a room may not fall on third-octave bands. With their variable-frequency capabilities, parametric equalizers let you tune right in on any problems that may exist in a room." [Another explanation of the unit's working is included in a feature article describing the Golden Nugget sound system, to be found elsewhere in this issue — *Editor*.]

Jaeger's console — a 32-channel Sound Workshop Series 34 — is located in the center of the control room, with the mix position 11 feet



A separate rack houses a pair of E-mu Systems voltage-controlled high- and lowpass filters (*top*), and three voltage-controlled phase shifters built by Wausach Music.

DIGITAL SAMPLING TECHNIQUES

Digital sampling is a powerful music making tool. However, as in every branch of audio, what you get out of it is only as good as what you put into it. Obtaining realistic orchestral sound samples is something of an art in itself. Based on his experience with sampling, Denny Jaeger offers the following advice, which is applicable to all high-quality sampling devices:

- "The longer the sample, the more you're hearing an actual performance on an actual instrument. That's the key to realism — capturing the unusual overtones and other nuances of particular instrument."
- "Don't take just one sample of anything; instead, take different passes for every single note you want to sample. If you're not adept at sampling, one of the five samples will probably be right. If you *are* adept at it, you'll have your choice of five different samples for each instrument."
- "Take samples bright — a little brighter than you would ever want to hear them in a composition. This preserves the overtone structures that make the sound of instruments unique. You can always roll brightness later without adding hiss."
- "Remember that each individual note you sample will later be assembled into a 'patch' on your digital synth, where it will be used to recreate a number of pitches apart from the pitch that originally was sampled. Digital sampling instruments work by varying the sampling rate used in encoding the original note. Thus, a sample of Middle-C on an instrument may be used to play the pitches ranging from the D immediately above Middle-C to the B-flat immediately below it before the next sample in the patch kicks in. This fact has several implications."
- "Whole steps are generally the closest intervals at which you should take samples. Half-step changes are imperceptible on virtually all instruments. For many instruments, a whole step or even a minor third will be imperceptible. For the majority of instruments, it's acceptable to sample every fourth. Even though changes may be perceptible if you listen to the patch in isolation, you will be using it in combination with other sounds in a piece of music. In that case, the canges will not be noticable."
- "If you are sampling every fourth or fifth note, know which fourths and fifth to sample in order to catch the true characteristics of your instrument. It may not be a good idea to take F-sharp and C-sharp on a B-flat trumpet, for example, as these are notoriously bad notes on that instrument; B-flats and F-naturals may sound better. Also watch out for dead spots and other quirks on the particular instrument you're working with. Work around them

DIGITAL SAMPLING TECHNIQUES — continued . . .

rather that plodding through mechanically and sampling every consecutive fourth or fifth, even if one happens to fall on a dead spot. Talk to the player. Ask him about good notes and bad notes on his particular instrument. On brass instruments, some notes are harder to keep in tune than others. You don't want to use that one bad note to represent four or five pitches in your patch."

• "Sax, clarinet and cello are especially hard to capture, because they each have enormously different tonal qualities, depending on which register you're playing in. Transitions between the different registers can be awkward when you're building patches. It is therefore best to sample whole steps or every note on these three instruments."

• "If you're sampling single instruments (as opposed to a section), have the player use no vibrato. This may take some effort in some cases, but it's worth it. Here's why: If you sample C4 on a flute with the normal vibrato a flautist would use, and then play it back an octave lower, the vibrato rate at that point will be half as fast as the original sample — which sounds extremely unnatural. On flute, it's also wise to have the player avoid using a lot of breath, since the breath will only turn up as hissy white noise and plague you later on."

• "With a string section, on the other hand, the more vibrato you can get the players to use, the richer the sound will be. (Also, the more players you use, the richer the sound will be.) But while you want a lot of vibrato, you also want the section to be collectively in tune. Striking just the right balance between richness and correct pitch is what makes string samples absolutely the toughest."

• "Because brass instruments can put out incredible SPLs, there can be distortion problems if you are putting your samples on tape before encoding them into your sampling machine. It's better to use a digital tape machine for these samples. This way, you can get better dynamic range while avoiding distortion."

• "With any wind instrument, give your players a brief warm-up and then start with your highest and hottest samples. If you try it the other way around and start with your lowest, quietest notes, the players may be too fatigued to hit the really high notes when the time comes."

• "If you're sampling the human voice, the best thing to do is take 'oooos,' 'ahhhhs' and 'eeees.' And you had better take every half step; four to six samples each. Otherwise, it will sound very mechanical and synthesized if you are using your vocal samples to build a choral passage. The result could be very beautiful, very unique, errie, etc.; but it won't be realistic." □□□

SYNTHESIZERS IN THE STUDIO

from the ACD monitors. The console offers four-band parametric EQ, 24 group busses and full ARMS Sound Workshop tape-based automation on all 32 VCA-equipped input faders. Jaeger is planning to upgrade to the SW Discmix automation system in the near future. By moving to a disc-based system, he will be gaining frame-accurate automation with a data transfer rate and storage capacity that far exceeds the capabilities of a tape-based system.

As with his choice of Meyer monitoring, Jaeger selected the Sound Workshop board because it seemed particularly suited to the needs of his work. "If you're doing a lot of digital synthesis," he comments, "You don't need a lot of talkback facilities and cue sends on your console. I'm working alone most of the time — once in a while I'll have a vocalist, a guitarist or maybe a few string players in the room. I don't need an elaborate cue system to deal with that."

Apart from this operational requirement, compactness and ease of use were also major considerations: "When you're doing music and engineering for yourself, the equipment had better not get in the way. The musician side of you is involved in a very frail decision-making process. If you get worn down or frustrated dealing with the equipment, you *won't* make creative decisions as well. The equipment has to be fast and easy to use, which is what I like about the Sound Workshop board.

"It's quite simple to set up group mutes and solos, subgroups, masters, etc. I can quickly solo a group of 12 tracks, say, and then solo any of the tracks within that solo just by pushing a button too.

"The board is pretty flexible as well. I can patch around the EQ or apply echo sents pre- or post-fader at will — all of the standard options are there. The only limitation is that I'd like to have more echo sends, because the way I work calls for a lot of effects. But, because I won't work with other musicians a lot, I've got six cue sends that I can use for echo."

The analog audio tape machines in Jaeger's facility are positioned along the left-hand wall, and include a Sony MCI JH-24 transformerless 24-track, JH-110 four- and two-tracks, plus an Otari MTR-90 Series II that is used as back-up. Jaeger has remote control of these machines at his console via a custom Control Video synchronizer/controller. Also available is a Nakamichi PCM processor, which he usually uses with a Sony SL-2700 Betamax VCR. The studio's principle videotape

. . . continued on page 89 —

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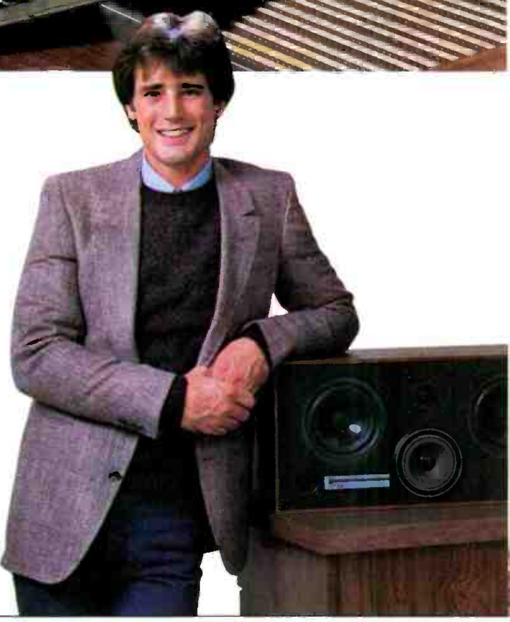

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June 1985 Number 5

ECHO TIMES

Digital Recording on Hard Disc – from the pioneers of Loop Editing.

THE STORY SO FAR...

A.M.S. (Advanced Music Systems) is a company well recognised for providing the professional audio and broadcast industries with one of the most comprehensive range of digital sound processing systems available. Within the product range already marketed by A.M.S. exist both hardware and software which have made possible the sampling, editing and transposition of audio material. These basic facilities have already had a significant impact on popular music production as well as on film and video post production. The attractions of being able to sample either live or prerecorded sounds digitally, then rapidly and easily to electronically edit that information are obvious. In popular music, any sample whether drum beats or even complete backing tracks, can be captured, edited and triggered on demand by such external sources as programmable music computers or simply by an audio input.
In the case of film or video, either the existing soundtrack or an events controller can be used to accurately synchronise a sampled and edited sound effect to the picture. These techniques are being employed now and are considered highly effective – in addition to providing significant time savings over conventional methods. This particular sampling technique, has been pioneered by the LES (Loop Editing System) on the A.M.S. DMX 15-80S.

MULTI-SAMPLE RESEARCH BEGINS

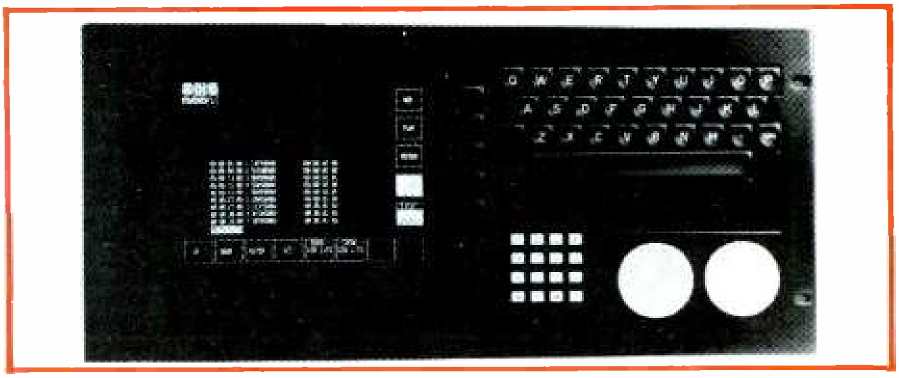
In late 1982 A.M.S. completed development on a Digital Preview Editor, the DMX 16E. This system was originally designed to work in conjunction with digital audio recorders to allow trial edits to be performed at variable speed, or by reel rocking, with programmable cross-fades.
These trial edits would be performed with the magnetic tape stationary.
The DMX 16E could support over 30 seconds of audio storage, and with the growing interest in LES experimental work was carried out on the system to investigate multiple sample storage and cataloguing. In June 1983 a highly modified DMX 16E was demonstrated to capture, edit and trigger multiple samples stimulating yet further interest in a product which was rapidly turning into a solid state digital audio recorder.

A.M.S AUDIOFILE IS BORN

AudioFile was shown in prototype form for the first time at the AES in Hamburg and the NAB in Las Vegas and is the result of over three years' research and development at A.M.S. into a hard disc based digital recording and playback system. AudioFile is capable of being configured in several different ways allowing it to perform completely different functions.
In its simplest form, AudioFile can capture samples of sound, edit those samples and store them in a non-volatile form for recall and playback at any point in the future. This is the first major difference from the DMX 15-80S where samples are lost on power-down.
Complete stores or files of sound effects can be recorded, edited, catalogued and saved within the AudioFile memory.
Secondly, AudioFile can have samples assigned to any of its outputs for multiple synchronous triggering. This triggering can be effected either manually, by audio input, by an events controller or by using AudioFile's built in SMPTE time code reader/generator. AudioFile can also be configured as a multitrack digital recorder – however, in this form it offers significant advantages over the conventional magnetic tape machine in that it is able to advance or retard any individual "track" with respect to any other. Tracks on a conventional multitrack recorder become digital files on AudioFile and the storage capability of AudioFile means hundreds of files may be stored at any one time and delivered to any one of AudioFile's outputs on cue.
AudioFile can run independently against its own internal clock or it can be locked via its timecode reader/generator to film, video or other magnetic tape recorders.

THE TIMECODE DIMENSION

A main advantage of using AudioFile in synchronism with any other machine is that the "elastic band" effect of having mechanical tape transports locked by a synchroniser is completely eliminated. On looping video, the audio will be heard in exact time with the picture virtually as soon as the video settles into play. Although the number of simultaneous tracks available in the "multitrack" form is limited at present, AudioFile with its eight existing outputs has many immediate applications.
Once the fundamental attractions of AudioFile have been accepted, it is possible to see that the system can "invisibly" provide additional digital audio "tracks" by locking it to a conventional analogue or digital recorder.
Alternatively, with specialised



software, AudioFile can eliminate the need for a multitrack recorder, synchroniser and desk automation system when track laying audio against video.

DIGITAL EDITING MADE EASY

Audio File can also be used as a digital stereo editing system. Because this editing is fully electronic and has inherent random access it can be conducted on a single machine with an accuracy of microseconds with totally unprecedented speed and flexibility.
AudioFile is working now and Winchester Disc Storage of audio is here to stay. A.M.S. with their successful range of digital audio processors have long understood the advantages to customers of updatable hardware and software – the absence of second hand A.M.S. units of any kind speaks for itself. A.M.S. are committed to making AudioFile the most versatile and upgradeable audio production workhorse available.

Specifications:

ELECTRICAL:

Digital Coding:	16 bit linear PCM
Sampling rate:	48kHz standard, (switchable 50/44.1/40kHz)
Frequency response:	20Hz to 20kHz (48kHz sampling)
Dynamic range:	Better than 90dB/ ref full output at 1kHz
Inputs:	10K electronically balanced
Outputs:	100R compensating electronically balanced

MECHANICAL:

Processor/first disk module:–	5U rack mounting
Disk Expansion module:–	5U rack mounting
Display/control surface:–	5U rack mounting

CONFIGURATION:
Inputs:
Modular/ 2 inputs per module
Outputs:
Modular/ 4 outputs per module

INTERFACES:
RS422 control for peripheral equipment
SMPTE reader/generator
Optional control for AMS audio processors (Reverb, etc.)
RS422 control of all AudioFile functions

CONTROLS:
High resolution (800 x 480 pixels) graphics display
13 function keys/ software definable
2 digipots for simple parameter setting/ reel rocking, etc.
Software/ application dependent starburst LED labelling for digipots
Built in alphanumeric pad for quick titling

CTS.! DSP.! AMS.!

CTS studios in London became fully operational with their Neve DSP console earlier this year. The console is digitally interfaced to the Sony PCM 3324 and as can be seen from the photograph opposite, CTS's choice of digital audio processors is A.M.S. When questioned as to why the DMX 15-80S and the RMX 16 ended up mounted in the DSP chief technical engineer Henry Edwards commented – "They are used all the time so it is the most natural place for them!"

PEOPLE IN THE KNOW

"If I walk into a studio and I don't see an A.M.S. digital reverb and an A.M.S. digital delay, I start having my suspicions about the place. I use the harmoniser function on the A.M.S. DMX 15-80S delay system, plus delays and I always use their reverb if I can't use natural reverberation. Other ones are good, but they always sound a bit plonky – they've got a sort of tone on them that I don't like. The A.M.S. one is superb – fully equalised top and bottom."

Colin Thurston, interviewed by Jane Angus in HSR Magazine.

On the effects side they bought a 6½ second A.M.S... They chose the A.M.S. because it was the only one which would do the job they wanted.
Tony: "The great thing about the A.M.S. is that it's here to stay."

Peter Buick of Sound Engineer Magazine talking to Tony McGrail of Terminal 24 Studios.

R-e/p (SB): I'm curious about one snare drum sound in particular: the title track of Springsteen's Born in the USA. *It has the impact of a .38-caliber revolver going off. How did you capture that?*
BC: What I did for a few songs on that album – and I think that was one of them – was to use the great sound I'd got from the stereo overhead mikes. The snare sound was amazing, for one thing because Max (Weinberg) tunes his drums really well. The snare drum mike itself wasn't happening, maybe because it was too close, but the overhead mikes were picking up this "Glyn Johns" kind of snare sound. So I just sampled that into an A.M.S., and it became the predominant snare drum sound, although it is mixed in with the original snare drum track. It was easy to do because there are no other drums playing during the intro part.
R-e/p (ML): So you're triggering that sound out of the A.M.S. DMX 15-80 for each snare beat?
BC: Yes, any signal you feed in will key it. You can also put little vocal snippets into the A.M.S. and key it off something like a bass drum. On the intro of the Hall and Oates album, there are some vocal bits, singing some Spanish words. By keying one off the bass drum and one off the snare, we have these little vocals answering each other exactly in time with the Linn Drum.

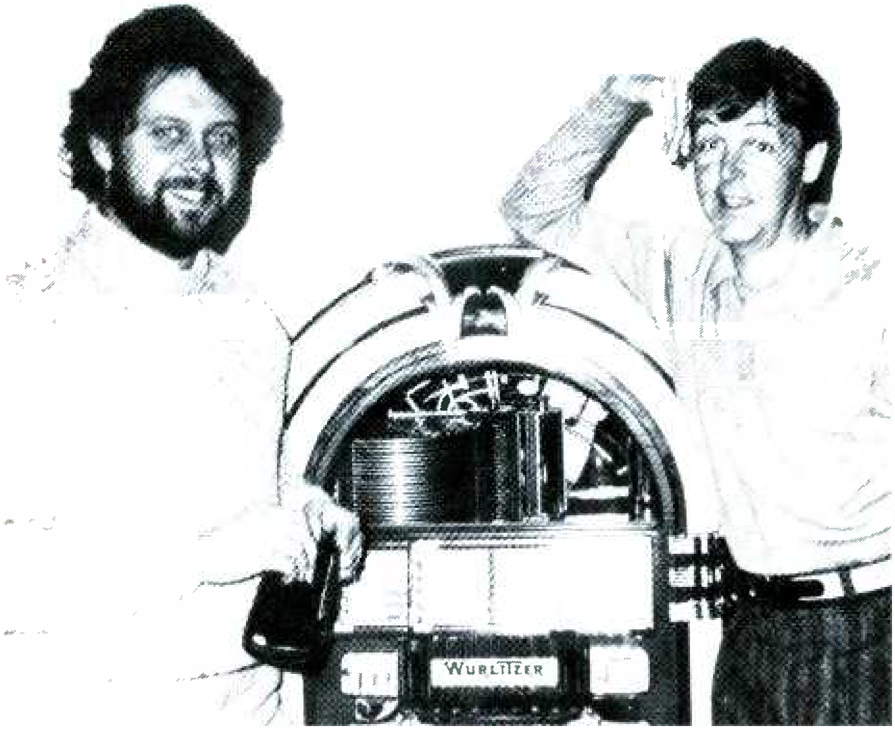
Bob Clearmountain, interviewed by Mel Lambert and Sam Borgerson for R-e/p.

I don't use any synthetic reverb at all if I can get by without it, but if I can't I'll use an A.M.S.

Colin Thurston, interviewed by Jane Angus in HSR Magazine.

A.M.S. would like to take this opportunity of passing on their sincere congratulations to Humberto Gatica on winning his recent Emmy. Humberto was interviewed in Echo Times No. 4 and we believe since then he has purchased a further RMX 16 digital reverberator.

Paul McCartney



Stuart Nevison of A.M.S., discussing another manufacturer's Audio Hard Disc System with Paul McCartney.

Paul McCartney needs no introduction, however, one of his lesser claims to fame is that he owned probably the third unit ever manufactured by A.M.S. Although Paul still maintains he is not technical I do believe that it is obvious from this interview that he understands how to get the best out of his A.M.S. units.

Paul McCartney now owns a DM 2-80, an RMX 16, and two DMX 15-80S systems. One DMX 15-80S is fitted with 14 seconds of delay and a keyboard interface whilst the second has 6.4 seconds. Both units have dual pitch changers and the de-glitch option as standard.

Paul McCartney: It's great though! It's fabulous! – We've just been talking about A.M.S. before you arrived, it's really a fairy tale in a way.

A.M.S.: Do you remember that first DM 2-20 Flanger?

P.M.: Oh yeh and I remember our first meeting at Abbey Road. We were very busy and I think you had explained that you had this Flanger that did this and that to a couple of the roadies. You were in studio two and I remember coming down the big stairs in 2 to see what was going on. I'm not technical and I'd just used equipment for what it would do but it did seem that the prototype you'd brought down looked and sounded good and had something together. So I thought the best I can do to give this fellow Northener an "in" is announce in a loud voice whilst other Abbey Road staff were around that I was having one and hope they'd do the same for you.

A.M.S.: Well, if we didn't say thank you at the time we had better say it now – because that certainly worked and I think EMI Abbey Road had the fourth DM 2-20.

P.M.: The next time I came across A.M.S. was when I realised people were using this thing and when I said whats that and got the answer – it's an A.M.S. – I said I know that don't I? Oh it was them!!! Much more recently I've got into 12" dance version singles and that's when it's become really interesting. When we had an original recording that lasted for 4 minutes but we needed 8 what we would do is invent something that could fit in, lock it in the A.M.S. and finally feed it in. We were creating new bits of tape with the A.M.S. It was great, what I would do is if I wanted to ad-lib a bit of blues singing over some chords I'd just have a mic run out into the studio and sing what I felt like and lock it into the unit – edit the sample and trigger it out wherever I wanted it in the mix.

A.M.S.: Did you use any of the other functions of the DMX 15-80S?

P.M.: That's another great thing you can do. By using the pitch changers you don't even have to have samples of the correct tempo. We lifted vocals from ballads that were made up of swimming big block harmonies that were completely the wrong tempo for where we wanted to drop them in, so we'd use the pitch changers in the 15-80S to correct pitch and then drop them in. You can even do that if you find that you've got a set of complicated harmonies early in a song that you just can't get exactly the same feel into later in the song. In fact I recently did that with an American producer on a complicated guitar part – the first verse was really where I'd hit it, other verses although good we both knew just weren't up to the first – so we lifted the first verse.

A.M.S.: That's using A.M.S. units as a pretty clever production tool but did you use it purely for effects at all?

P.M.: Oh yeh, I had a lot of fun using timecode and noise gates to trigger edited samples from the unit. That can be used to do something that people have always been doing in pop music production and the "hit record scene" – and that is generating something that catches your attention. The sort of thing Trevor Horn is doing now, the Beatles did ages ago the difference being that in the early days we did it with rubber bands and sealing wax and now as we move further into the computer age a lot more possibilities exist.

A.M.S.: So how do you see A.M.S. systems fitting in?

P.M.: In truth George Martin and I have just worked on three albums together – Tug of War, Pipes of Peace and Give My Regards To Broadstreet. The last thing we did out of all that recording was the 12" single of No More Lonely Nights. What was really great was that working with A.M.S. units I certainly felt as excited with the kind of possibilities as we had in the very early Beatle days because we thought anything was possible. We used a lot of sampling, resampling and even sampling material from other tracks. I think it is fair to say that out of all those three pieces of work I found that 12" version the most exciting and that did coincide with the fact that we got most heavily into the A.M.S. on that particular single ■

Not too many years ago Hugh Padgham was driving a van for a hardware store when he made a delivery to what is now Farmyard Studios in Little Chalfont. To say this delivery and chance meeting with Rupert Hine and Trevor Morace of Farmyard changed Hugh's life is something of an understatement. Hugh Padgham's career has now been in full swing for quite some time and his mark has been left on many of rock's major works, including material by Peter Gabriel, XTC, David Bowie, Hall and Oates, The Police, Genesis and Phil Collins. As this article is being written Hugh's most recent piece of production is at number 1 in the UK album charts – No Jacket Required, by Phil Collins.

Hugh Padgham: My first job was as a tape op at Advision Studios in 1973/1974. I really didn't last long and I was laid off for being incompetent – but that was fair enough because there just wasn't anybody to teach me the ropes. I then got a job at Lansdowne Studios where I stayed for 5 years and looking back that period was very beneficial for me because I got involved in every sort of recording – from Jingles to Jazz, pop, rock, strings and even small orchestras. The team at Lansdowne were fastidious in their approach to recording and that definitely rubbed off on me. In my summer holidays I did live sound on tour for Rupert and Trevor's band Quantum Jump and that was a lot of fun. I eventually got a job at Townhouse during its very early days – I always feel very good about that late 70's period when the Townhouse had the first SSL in London and probably some of the first A.M.S. units as well.

A.M.S.: So was the Townhouse the first place that you came across any A.M.S. equipment?

H.P.: Yes it was. This was even before the DMX 15-80S had been introduced and I can remember using the mono DMX 15-80. That very unit is still sitting in the rack at Townhouse and I still use it. I love the regeneration filter that was on the old mono units. For me, probably one of my most exciting uses of an A.M.S. delay line was during sessions with Hall and Oates on the Maneater track. There was a sax solo in the middle that I didn't like – there was the odd sax phrase and then a huge gap till the next phrase. I thought, I know how to sort this – Electric Lady had a good amount of delay in their A.M.S. so I fiddled around with the first sax phrase and got it to repeat in the gap. I think the result was really great.

A.M.S.: Is it possible to say exactly what it is you like about the DMX 15-80S?

H.P.: What's so wonderful about using an A.M.S. is you do get out exactly what you put in. On the new Phil Collins album most of the vocals employ the 15-80S with a 1.007 pitch change and an 11 or 12 millisecond delay – it's clean and it really suits his voice. The unit is brilliant for turning something mono into a big fat stereo sound. Again with reference to the No Jacket Required album we took the Earth Wind and Fire horn section and placed the original sound in the centre and the two outputs of the DMX 15-80S panned left and right. You end up with a really fat sound with the same quality of sound in the centre and on the right and left – you can't do that with any other system. I must admit, it is horses for courses and I can still find uses for tape delay when even analog degeneration can be appealing.

A.M.S.: What are your feelings about the RMX 16?

H.P.: What I love about A.M.S., and also SSL, is they do seriously consider the "art" side of the business and it isn't just white coated people inventing a piece of equipment that they haven't a clue personally how to use. It is refreshing to feel that there are people genuinely interested in the art and I remember a phone conversation with A.M.S. who quizzed me about the now legendary "Intruder" drum sound on Peter Gabriel 111. The result of that call was obviously the Nonlin program in the RMX 16. I would say A.M.S. has influenced modern day music to a very great extent with the RMX 16 – particularly with the Nonlin and Reverse programs. Those programs are so recognisable and you hear them everywhere – a lot of people would be very lost now without an RMX 16 I can tell you!

A.M.S.: Does that mean you'd be lost without an RMX 16?

H.P.: I certainly couldn't do a session without A.M.S. units! I couldn't walk into a studio if they didn't have A.M.S. – it would be like someone taking off one of your arms! A.M.S. have undoubtedly changed the face of modern music.

— *continued from page 82 . . .*

SYNTHESIZERS IN THE STUDIO

machine is a JVC CR-6650-U ¾-inch U-Matic, which is located in the tape machine library.

Apart from the console and analog tape machines, virtually all other control-room equipment is mounted in a moveable console, and can easily be located anywhere behind the mix position. The Synclavier keyboard is set up on a rolling stand, and the effects rack is fitted with casters. Outboard equipment housed in the effects rack includes a Quantec Room Simulator, Lexicon Super Prime Time, Lexicon 224X digital reverb, Symetrix compressor/limiter/expander/gate-ducker, Symetrix Peak-RMS compressor/limiter, UREI LA-4A compressor/limiter, UREI Model 537 graphic EQ and Model 539 room EQ (which is used for outboard equalization only).

Inputs and outputs to every tape machine and every piece of processing gear appear on a patchbay mounted to the right of the console. This is supplemented by a video patchbay also mounted in the effects rack — that connects to every video deck in the studio. While providing Jaeger with ready access to any picture source, the patchbay enables him to encode digital audio, via the Nakamichi processor, onto any of his video decks.

"By having all my equipment appear on one of the two patchbays, I never have to run around cabling anything," he comments. "I want to work very fast and very clean; this system lets me do that. I can roll my keyboard anywhere behind the console, and grab for my effects rack."

Recording Techniques For Digital Synthesis

"I was trained in traditional recording techniques," Jaeger asserts, "but many of them just don't seem to hold water when it comes to recording digital synthesis. It doesn't really matter which digital synthesizer you are working with either — they *all* have very fast slew rates. Also, you get extremely concentrated energy at certain frequencies.

"The challenge [of working with digital synthesis] is to get all of this down on tape without any distortion. Some of my recording methods seem a bit abnormal by conventional standards; but, over the years, I've found them to be best suited to what I'm doing."

In connecting the Synclavier's audio outputs to his console, Jaeger comes in on the microphone rather than the line inputs. He finds that the mike inputs are better able to handle the wide range of voltages produced

The studio's Sound Workshop Series 34 console and Control Video timecode unit (*above*). An outboard rack houses (top to bottom): a Sony PCM-F1 and Betamax VCR; Quantec Room Simulator; Lexicon Prime Time; Symetrix compressor-limiters; UREI Model 537 and 539 equalizers; plus various Audio+Design Scamp modules.

by the synthesizers. "A lot of the sounds that come out of the Synclavier have enormous dynamic range," he explains. "The signal may be a fraction of a volt at one moment, and then go as high as seven volts. If you go into a line input with something like that, some of the sounds are going to start getting a little noisy; others are going to be ridiculously hot. By going into the microphone inputs, I can take advantage of the console's 20-dB switchable pad. I use it in conjunction with a 15-dB switchable pad on the Synclavier's output to get the least possible noise."

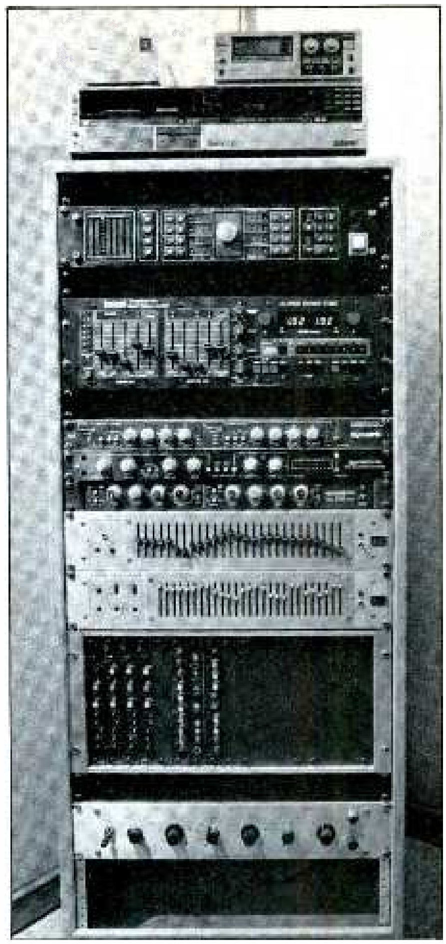
Jaeger has developed the practice of rebiasing specific tracks on his multitrack, depending on what type of Synclavier sound he is printing.

"In the past," he recounts, "I noticed I would take very heavy transient sounds — spikey plucked piano or vibraphone sounds, for example — print them at levels as low as -10 or 20, and *still* sometimes get distortion. I found that by changing the bias, I could get around that problem.

"Bias is basically a trade off between how much high frequency you get on tape, and how little distortion you create I sometimes overbias, sacrificing a little high-end for better distortion characteristics.

"The thing about synthesis is that you can alter the frequency content of the sound source itself. It's not like recording a conventional instrument, the tonal characteristics of which are an unalterable given. So while I sacrifice a little high-end by overbiasing, I can add a little more apparent high-end energy to the sound itself.

"I use Ampex 456 Grand Master tape, and always roll at 30 ips. I align the machine at +7 using the standard 250 nW/m reference. Now, the recommended amount of overbiasing for Ampex 456 is from 1.5 to 1.75 dB at 10



kHz. But I will frequently go as much as 2 or 2.25 dB over at 10 kHz for bright transient sounds such as snare drum, any type of metallic percussion, vibraphone, plucked piano or even harp simulations. For timbres like violins or brass, I may go 2 dB over — always at 10 kHz, 30 ips."

One drawback to printing tracks at elevated levels is print through, which Jaeger will often allow to occur and then mute during mixdown using the Sound Workshop automation. Print through occurs most frequently when a very loud passage — such as a brass hit, for example — occurs very suddenly, and then ceases just as abruptly. Jaeger is generally able to mute such tracks right up until the onset of the loud passage, and then mute them again after they cease. This procedure is generally sufficient to clean up any print through that there might be on a multitrack master.

"Print through isn't so severe that it will go from one track to another," he points out. "If you have a big brass hit on track #5, it usually won't print through to track #2; you'll get all of your problems on track #5. If your computerized automation is fast enough, you can remove the print through image on that track. Also, the technique of muting tracks whenever there is no music on them lets me pick up as much as 10 dB in signal-to-noise ratio in my final mix."

AMS FOUR-PAGE INSERT PRECEDES ON PAGES 85 to 88

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Jaeger prints most of his effects during tracking, rather than adding them during the mix. He tends to use outboard effects and processing devices as an extention of the synthesis process itself. "What I put on tape during tracking is as close to the finished sound as I can get. I view things like DDL programs, flanging, chorusing and phase-shift programs as being *part* of the sound I'm creating. On the [Lexicon] 224X, for example, they have a squarewave setting that you can set for a very fast rate, and create a flutter-type of effect. I'll use that on synthesized flute or trombone tracks to get a little of the 'grit' associated with those instruments."

Along with his principle effects rack, Jaeger also has a small rack of "custom goodies" that sits beneath his console, and houses the devices which can respond to control voltages from the Synclavier. Among them is a set of three voltage-controlled phase shifters built by Wausach Music, and later customized to Jaeger's specifications. The three can be ganged together and controlled — individually or collectively — by external signals.

"There are separate cue controls for each of the three," Jaeger says, "so I can emphasize certain frequencies and feed them back, as well as control the waveform that's sweeping the frequencies. What I usually use in these phase shifters is a waveform that's a cross between a sine and a triangle wave. A plain sinewave spends too much time in its crest and trough, so there is a period where the phasing effect seems to cut out. A triangle is actually a much better waveform to create a phase shift, because it doesn't waste any time in its upper and lower extremities. But sometimes it's a little *too* sharp, which is why I go for a modified, or softened, triangle shape."

Mounted above the bank of phase shifters is a set of E-mu Systems voltage-controlled high- and low-pass filters. Like the custom phase shifters, Jaeger has been using the E-Mu units for some 12 years. "I'd have to say that they are the finest filters I've ever used. I have hung onto them because they're *ultra*-clean. What I do is 'play' the filters by hooking them up to Synclavier's velocity- and pressure-sensitive keyboard. But often I'll operate them manually to get filter shapes that you just can't achieve using the keyboard as a controller. I can also gang the filters together, so that when I turn just one knob they operate as a bandpass filter, enabling me to roll off highs and lows simultaneously"

Synchronization Facilities

The timecode synchronizer used by Jaeger is a Control Video editing system. Working closely with the company, Jaeger has modified the device to also work with all of his audio machines. The system is still in prototype form and, at the time of *R-e/p's* visit, the synchronizer's compact control unit (approximately four by seven by 11 inches) was still housed in a make shift cardboard chassis. The system works by locking each machine to its internal 59.94 Hz crystal-derived reference, a process that Jaeger finds infinitely preferable to one in which individual transports are slaved to other machines.

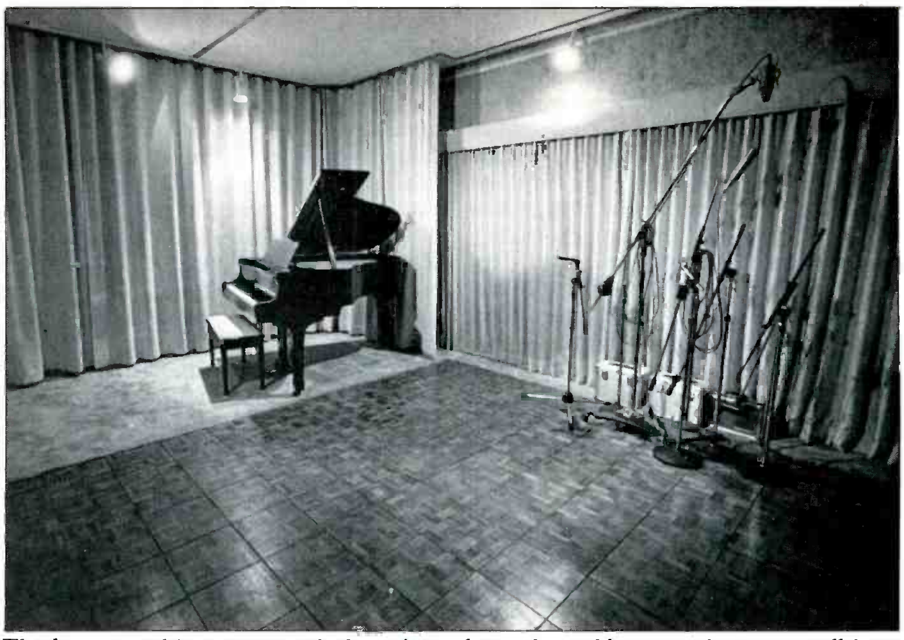
"This way," he explains, "if one of the machines runs into a little problem and goes off speed, it doesn't affect any of the other machines that are locked up."

The system also affords Jaeger the kind of operational simplicity he needs for his one-man working situation. He can advance or rewind the video and audio simultaneously, without having to take his eyes off the control room's Sony Pro Feel Video monitor. A ribbon controller on the unit's front panel enables him to roll forward or backward one video frame at a time, or to fastwind continuously in either direction. A cue function enables all machines to be rewound to any designated frame number, and the panel also includes remote-control facilities for all record functions that can be assigned to any audio or video machine in the studio.

The Control Video synchronizer can read and write any SMPTE/EBU timecode format. One feature of the device that Jaeger finds quite useful is the ease with which its SMPTE timecode reader can be slaved to its SMPTE writing facilities. "Very often," he explains, "I get a ¾-inch video master from a network or movie studio which will already have SMPTE on the address track. In a situation like that, you want to be able to lock a 24-track to the video master in order to write your music to picture. The easiest way to do that is to ask your synchronizer to read the SMPTE onto the 24-track, programming in the necessary one-frame offset between the reader and writer. You can then read the timecode off the 24-track, and it will be in sync with the picture.

"It sounds simple enough but, on other synchronizers, I've found that you have to type in a million commands to slave the reader to the writer. With the Control Video synchronizer I can achieve the same thing with two buttons."

Jaeger prints his click track, timecode, video reference frequency (if any) and automation data on tracks #1 thru #4 of the multitrack. The timecode and click tracks are printed at -20 VU, while the console automation data goes to tape at -7 VU. Jaeger then begins recording his music on track #24, working downwards to lower numbered tracks; in this way he avoids crosstalk bleed from the click and code tracks.



The large tracking room includes triangular redwood louvers that cover all four walls to provide non-parallel surfaces. Absorbent drapes can also be pulled across the walls to further deaden the area's acoustics.

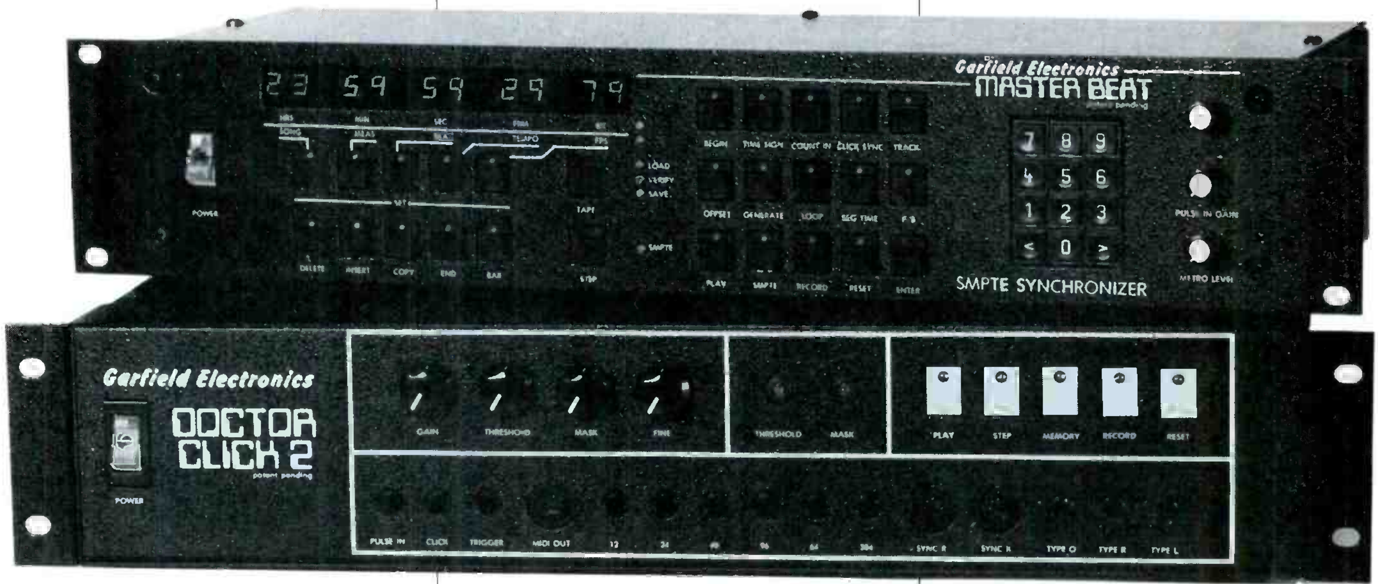
Thanks to a recent update made to the Synlavier, which adds SMPTE timecode capablities to the system, it is now a simple matter for Jaeger to lock the synthesizer's 32-track digital memory recorder to timecode on tape.

"In most cases," he explains, "the Synclavier will lock to a tape machine that has SMPTE on one of its tracks. The way it works is that the digital synthesizer will wait until the machine is within 20% of speed, and then it instantaneously locks to the machine. In my studio though, the Synclavier will lock up directly to the crystal-oscillator clock in the Control Video synchronizer."

Even before SMPTE-reading facilities were added to the Synclavier, however, Jaeger was able to lock the system to virtually any kind of sync input on tape. "The Synclavier will sync to anything, although it prefers to see five volts with a 10-millisecond pulsewidth. For a long time I used to use a little device, made for me by Serge Electronics, which takes any kind of an input and puts out 5V at a 10-millisecond pulsewidth.

"NED has since modified the Synclavier so that it will clean up sync input signals by itself, so I rarey use the Serge Electronics device now. But if I have a signal where the pulses have long trailing edges [i.e., signals with a long, gradual decay time] I will put it through a noise gate to get rid to the trailing edges. I'll then put the signal through a line amp, and adjust the gain so that the Synclavier is reading it perfectly. By adjusting the pulse width with the gate, and the volume with the line amp, I can make the Synclavier track anything . . . dog

CLICK to the BEAT



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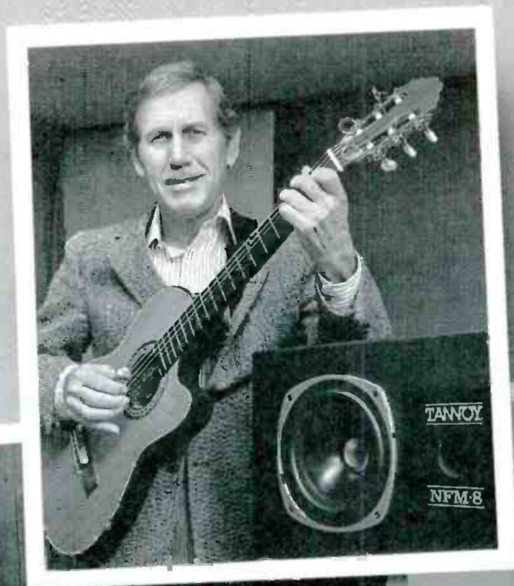

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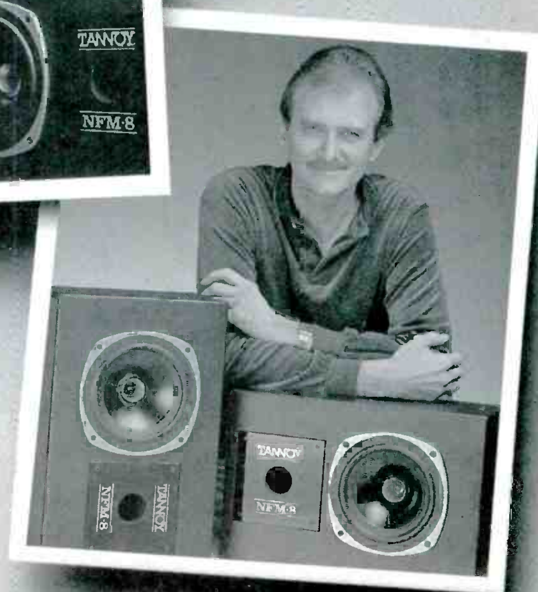

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barks, you name it."

For this reason, if Jaeger is using a drum machine or synthesizer with the Synclavier, he will synchronize the latter to the external synth rather than using it as the master clocking device. Although his studio is equipped with a Roland Jupiter 8, a Sequential Prophet T-8 and an Oberheim DMX drum machine, Jaeger finds that he has come to rely on the Synclavier more or less exclusively for all his synthesized sounds.

Future Expansions

The next step Jaeger has planned for his digital synthesis studio is to upgrade the data-storage system. "I've been exploring the idea of a storage system for the Synclavier that will enable it to use an Optical Memory Disc Recorder [OMDR], such as the ones made by Thompson CSF or Data General. I'm designing software and a handshaking device to get this type of storage system to 'talk' to the Synclavier. An OMDR system would enable you to store up to a gigabyte of data on one side of a disk — that's two gigabytes per disk. Right now, all you can store is 20 megabytes on the Winchester.

"My main idea is to put a library of sounds onto optical disk so I don't have to be constantly transferring thing, back and forth on and off the Winchester; it would be much less time-consuming than my present set up, because I could access any spot on the gigadisk in seconds. At present, my sound library is on hundreds of audio tapes and close to 45 Kennedy tapes. It's unwieldy to have to play librarian and locate a tape on a shelf every time I want to find a sound. With the new system — which I hope to have by late '85 — I can have my entire library on one or two optical disks. Beyond that, I'm also considering the possibilities of mastering to optical disk."

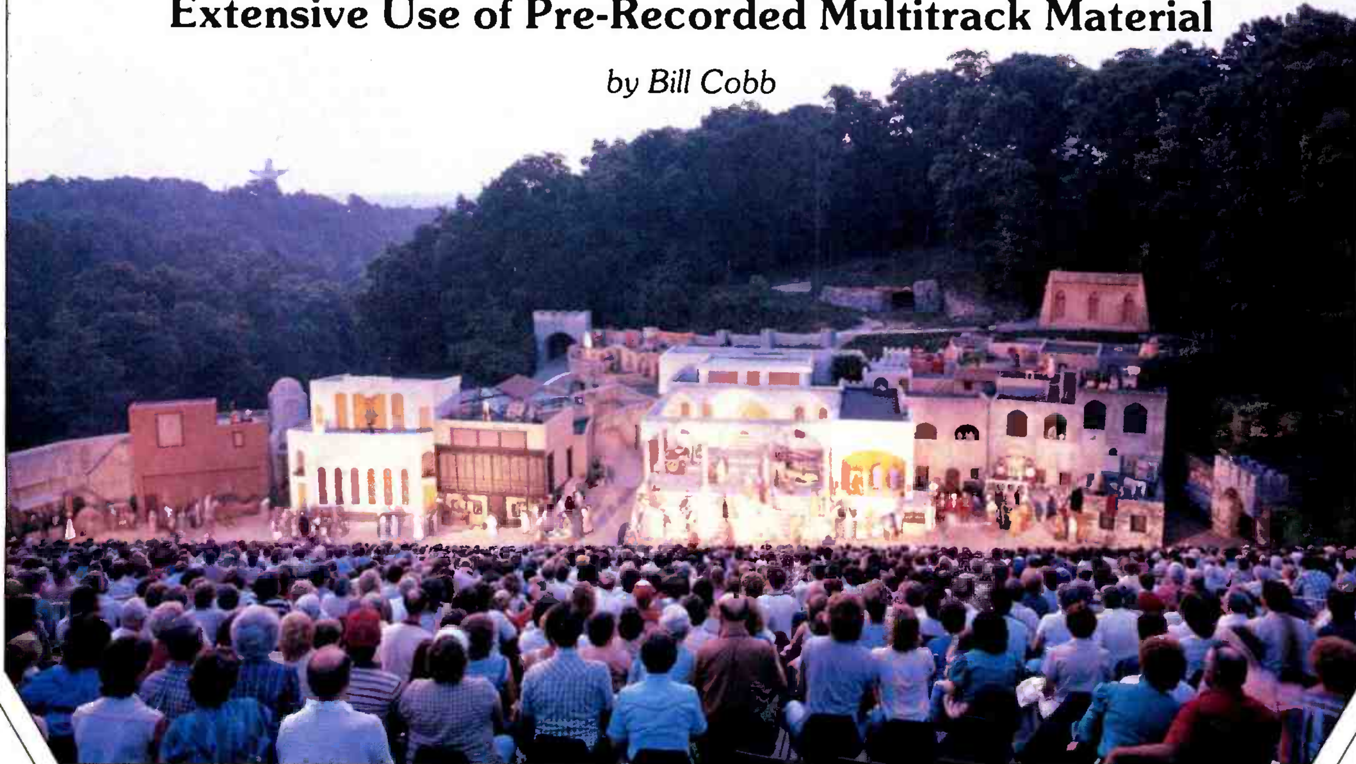
* * *

In making the one-man composer/orchestra/producer/engineer a reality, digital synthesis has had a considerable impact on the art and industry of film and television scoring. The potential of digital synthesis for emulating orchestral timbres and for creating unique new sounds has grown dramatically in the past few years. But, as this tour through one digital synthesis facility has repeatedly shown, technological advances that streamline the composition *and* recording processes are of equal importance. It is advances of this nature that will make facilities like Denny Jaeger's an increasingly viable and popular source of new musical ideas in the coming years. ■■■

MULTITRACK MUSIC AND VOCAL RECORDING FOR "THE GREAT PASSION PLAY"

A Live-Performance Production Involving the Extensive Use of Pre-Recorded Multitrack Material

by Bill Cobb



All photography by Bill Cobb

Throughout the course of one's engineering endeavors, occasionally you are confronted with a project so unusual that each phase of the project must be planned with great care and organization. Recently this writer was asked by a producer friend, Terry Talbot, to be a part of such a project. The task ahead was to completely redo all of the sound effects, music, and vocal recording for an outdoor stage drama called *The Great Passion Play*, performed each year in Eureka Springs, Arkansas. Before getting on to describe what all this recording entailed, a word of explanation about the play and its unique location would be in order.

The *Passion Play* was started 16 years ago with the goal of presenting the story of Christ's last days on Earth, from his entrance into Jerusalem on Palm Sunday, through his ascension, in such a way that the audience would feel that they had actually been there and witnessed it.

The show is performed on what is referred to as the largest outdoor drama set in the United States; it measures 500 feet wide by 300 feet deep, with a vertical rise from downstage to upstage of 55 feet. The *Passion Play* sees an audience of about 2,000 to 3,000 people per night, five nights a week, with its season running from April through October. (Last year the shows attracted just under a quarter of a million attendees, with up to 330,000 expected this year.)

The performance takes place in the small Ozark Mountain town of Eureka Springs, Arkansas, with a population of about 2,000; during the summer months, however, between 20,000 and 50,000 people a week come through the city.

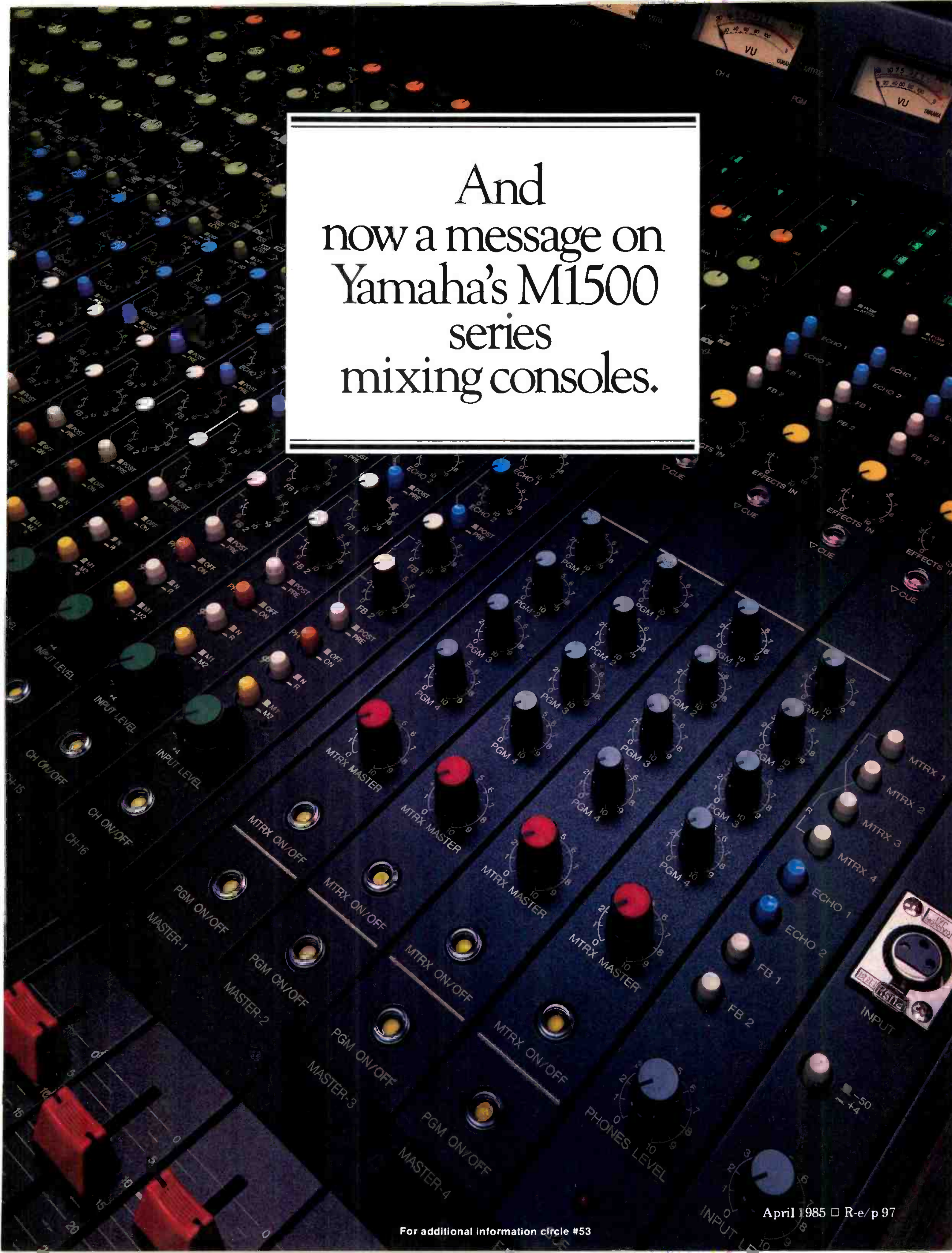
Within the last year, it was decided that there was a need to technically expand the show after 16 years of having basically the same look. The planned upgrading included some rebuilding of the set, complete electrical rewiring, installation of a computerized lighting system, addition of special visual effects, complete revamping of the sound system, installation of a 16-track tape machine, rewriting of the script, and re-recording of the complete soundtrack to include all vocal parts, effects and music score.

Overall technical direction for the expansion was being provided by Scott Bokowski, of Paragon Lighting, Newbury Park, California. Poiema Studios, with myself as engineer, was called upon by the producer to provide facilities for re-recording and mixing of the show.

At first glance this appeared not to be too complicated a task; however, as the project unfolded, things became *very* involved. Due to the uniqueness of the project, we were given special constraints to which we had to adhere, including:

- Sound coming from the stage set must give the illusion of coming from the actor's location.
- Movement during action scenes must be accurately reflected with respect to sound positioning.
- There must be no mixing of sound occurring during the actual performance; in other words, all vocal levels, effects, sound positionings and movements must be *premixed* to preclude possible mixing errors during the performance.
- All vocal parts must be recorded in such a way to allow ease of changing vocal lines and substituting new actors for certain future performances.

As may readily be appreciated, the above criteria added something of a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−128dBm Equivalent Input Noise (EIN);
−95dB residual output noise with all Faders down.
−73dB PROGRAM OUT (77dB S/N); Master Fader at nominal level & all Input Faders down.
−64dB PROGRAM OUT (68dB S/N); Master Fader and one Input Fader at nominal level.
−73dB MATRIX OUT; Matrix Mix and Master controls at maximum, one PGM Master Fader at nominal level, and all Input Faders down.
−64dB MATRIX OUT (68dB S/N); Matrix Mix and Master controls at maximum, one PGM Master Fader and one Input Fader at nominal level.
−70dB FB or ECHO OUT; Master level control at nominal level and all FB or ECHO mix controls at minimum level. (Pre/Post Sw. @ PRE.)
−64dB FB or ECHO OUT (68dB S/N); Master level control and one FB or ECHO mix control at nominal level. (Pre/Post Sw. @ PRE.)

MAXIMUM VOLTAGE GAIN (Input Selectors set at "−60" where applicable)
PROGRAM & MATRIX 84dB; Channel In to the corresponding output. EFFECTS 20dB; Effects In to PGM Out.
FB & ECHO 94dB; Channel In to FB/ECHO Out. SUB IN 10dB; Sub In to PGM Out.

EQUALIZATION (±15dB maximum)
LOW: 50, 100, 200, 350, 500Hz, shelving. HIGH MID: 1.2, 2, 3.5, 5.7kHz, peaking.
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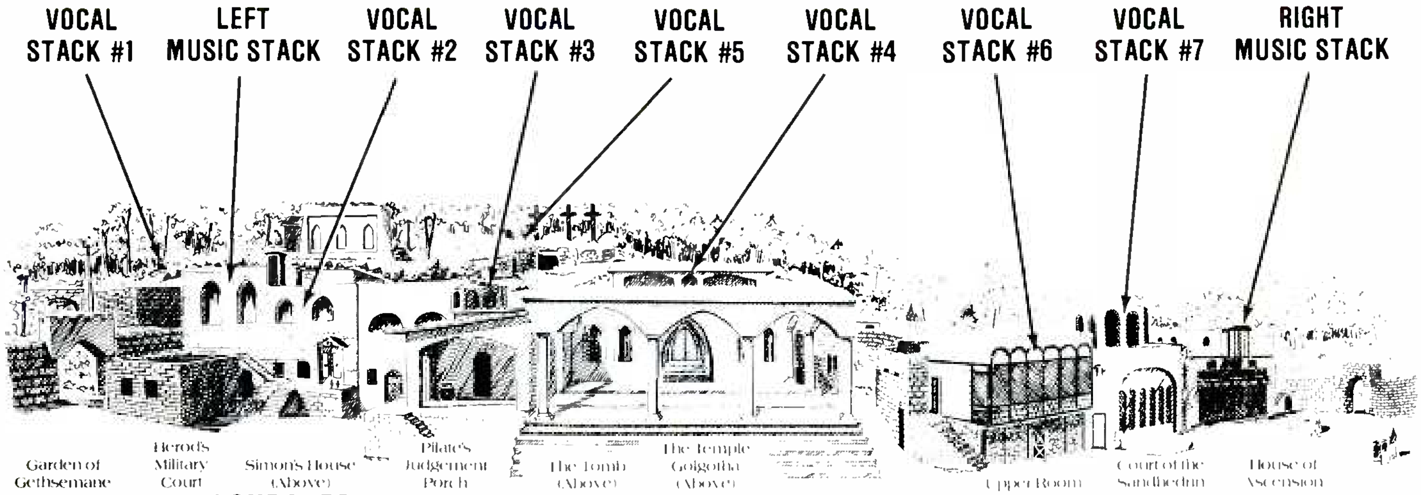


LOUDSPEAKER LAYOUT AND ASSIGNMENTS FOR "THE GREAT PASSION PLAY"

challenge to the recording project.

Sound System Layout

It was decided to dedicate seven speaker stacks located throughout the stage set to the vocal lines. Such a layout provided sufficient flexibility in sound placement to create the illusion of sound coming from the actors' exact locations, even though during certain sequences the action is taking place as much as 400 feet away from the audience front row. The loudspeaker stacks consist of Altec-Lansing two-way systems powered by Crown D150 amplifiers.

To handle replay of music during a performance, two music stacks located on each side of the stage set were added. Each stack consists of Martin bass bins and low-mid horns on JBL drivers, CLS high-mid horns on JBL drivers, and Emilar HF tweeters and drivers. Signal processing for the music stacks included Klark-Teknik graphic equalizers and Ashley compressors.

The complexity of actor movement within the live performance, and the need for the corresponding sound to move from stack to stack, dictated some form of automated mixing to eliminate the chance for human error. It was decided to use a 16-track with nine of the pre-recorded tracks directly feeding the seven vocal and two music stacks. In this way the whole performance would be replayed from premixed reels of two-inch tape. Use of a 16-track also left tracks open for control information needed to automate the computerized lighting system, future foreign language translations, and a separate track to feed monitor speakers located throughout the stage set. How this final two-inch master tape came into being, and the associated obstacles that had to be overcome, will be the subject of this article.

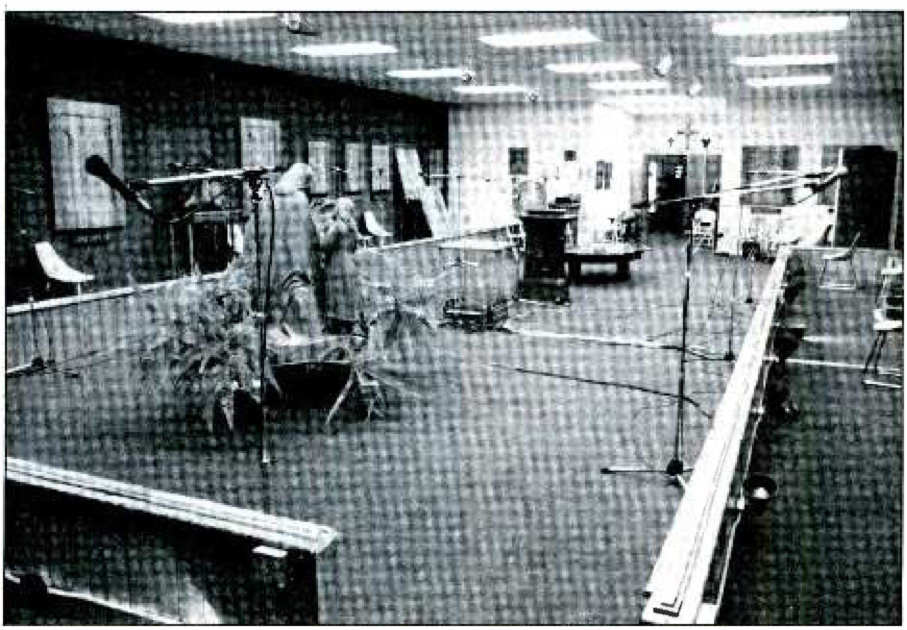
Vocal Recordings

The first step in the recording process was to record all the vocal parts. Since most of the actors recruited for the performance of the vocal lines were from the Eureka Springs area in Arkansas, we had to go on location to the site of the *Passion Play*. To provide location recording facilities, we recruited Web Staunton with his Mountain Mobile recording bus, which is based in Tulelake, California, just north of San Francisco. The vehicle was well equipped with an Otari MTR-90 24-track, Neotek Series Two console, UREI Time-Align monitors, and other assorted goodies to aid in the recording process; we were also impressed with the company's dedication to service, and interest in our project.

The first thing was to locate a building on the *Passion Play* premises suitable for vocal recording. The building we found proved very satisfactory, being relatively quiet with sufficient room for isolation between actors, while providing good access for the Mountain Mobile. The set-up consisted of 12 microphones, located around the perimeter of the room, with the director's microphone located in the center.

At this point, a certain amount of thought had to be given to the constraints under which we were working, in order to determine the techniques to be used for vocal recording.

Music recordings were made at CBS Studios, London, during nine, three-hour sessions extending over five days. The specially composed underscore written by Phil Perkins was recorded by a 45-piece orchestra.



— the Author —
Working his way through school playing in rock bands, **Bill Cobb** received a bachelor of science degree in engineering from the University of Florida. Upon graduation, he moved out to California where he designed and built Poiema Studios 24-track facility. He has functioned as owner and chief engineer since the studio opened in 1977.

As previously mentioned, the format of the final master 16-track is seven tracks feeding speaker arrays that correspond to the actors' location at any given time on the stage set. Because staging, timing, and relative level information was not available to us during the recording sessions, the vocals could not be recorded directly to these tracks. Instead, I decided to assemble a master two-inch multitrack tape, with each of the 12 vocal microphones assigned to its own tracks.

Once staging and timing information had been established, these tracks would then be remixed to provide vocal tracks for the final master. Also, because each vocal was on its own track, independent of the others, the flexibility of being able to replace lines with different actors at a future date was assured. In addition, the final vocal tracks would be mixed at their proper level and positioned relative to the actors' positions and movements on the stage set, thus precluding any live mixing during the performance.

For the next three weeks I more or less lived in the Mountain Mobile recording bus, recording the vocals and editing/assembling the separate performances onto the two-inch vocal master. We had to cover about 56 scenes, each recorded live with all actors in the scene at once, and many scenes having multiple takes. Due to the actor's availability and efficiency in recording, most of the scenes were recorded out of order, which definitely didn't make the arduous task of editing and assembling the performance much fun.

Following the week-long vocal session, for the next two weeks the director and script writer, Tom Jones, and I essentially buried ourselves in the bus, sequencing, editing, and determining timing between scenes and allowing for proper staging. We ended up with a two-hour show recorded at 15 ips on four reels of two-inch tape. (We chose 15 ips to minimize the number of reel changes during the performance.)

Which completed my assignment in Arkansas for the time being, except for one thing: we still needed crowd responses in some of the scenes, and planned a session with as many people we could still get a hold of to make up our crowd. Because all the crowd scenes take place in an outdoor setting, we couldn't use our room set-up; therefore, we just put our crowd right outside the bus. Everything went fine, just so long as no cars were going by!

Having completed the vocal and crowd recording phase of the project, we said goodbye to the fine folks from Mountain Mobile and headed back to Poiema Studios, Camarillo, California, for the next phase of the project.

From left to right: author/engineer Bill Cobb, composer Phil Perkins, and engineer Mike Ross-Trevor during music playback in the MCI-equipped control room at CBS Studios, London

Upon arriving at Poiema Studios, producer Terry Talbot and I began working on the next step, which was to record all special effects and background sounds necessary onto open tracks of the two-inch vocal master. This stage had to be done prior to remixing the vocals, because effects and background sounds had to be incorporated with the vocals onto the seven tracks of the final multitrack master. We were aided in this process by the help of Rhett Lawrence and his Fairlight CMI digital synthesizer. Rhett had many of the sounds we needed already stored on floppy disk, and we were able to create on the CMI most of the others that we needed.

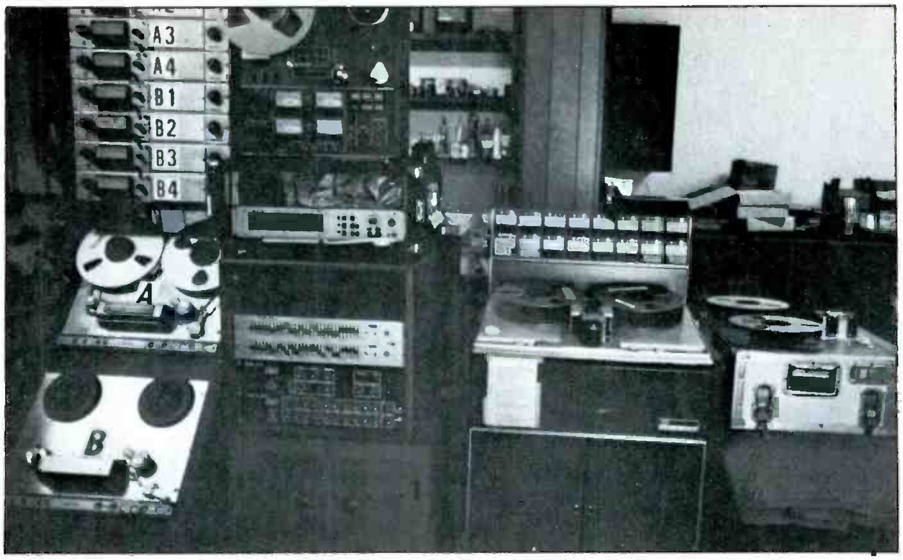
Master Vocal and Effects Remix

To facilitate mixdown of our effects and vocal tracks to the seven-track performance master, we called upon the folks at Recording Services Company, based in Los Angeles, to bring out a second multitrack (an Ampex MM-1200) and BTX Shadow SMPTE synchronizer, which they did with much grace. Timecode was laid onto an open track of the vocal master, and then transferred over to a blank reel of two-inch tape laced up on Poiema's Stephens 821B multitrack. The two machines were then locked up with the Shadow at the start of the first vocal tape.

Synchronizing multitracks in this way accomplished two things: Firstly, it enabled us to mix each half-hour tape scene by scene, and punch-in individual vocal lines to get the mix just right; and secondly, it facilitated future replacement of actors' lines, by first recording the new line in its proper place on the vocal master, syncing the vocal master with the final performance master, and then transferring that line over to the latter.

The task of mixing down the vocal master presented its own set of complexities. In the studio we had to set up a monitoring layout that would duplicate the stage set environment and speaker-stack locations. Poiema Studios is equipped with a Soundcraft Series 2400 console which, with its split design and versatile subgrouping/monitoring capabilities, greatly simplified the task. The 12 vocal tracks from the vocal master, as well as the special-effects tracks, were brought up through input strips and assigned, in accordance with information provided by the play's director, to the seven vocal "speaker stack" tracks (tracks 2 through 8) of the final master performance tape.

Room layout and general microphone setups used during the five-day schedule of underscore recording sessions at CBS Studios, London.



The Passion Play control room, with the 3M M56 used to replay the show's multitrack music, dialog, and effects tapes. To provide speaker stack assignments, a portable Stephens multitrack also was used to remix the music tapes.

The studio monitoring was set up such that all monitor faders for the seven tracks were set at the same nominal level, and their panpots positioned in the stereo soundfield at the same image location corresponding to the way sound would be perceived from the center of the audience seating area. Keeping the monitor faders fixed, once a proper monitor level was established, ensured that all level changes made during mixing due to scene dynamics, actor involvement, etc., would be accurately reflected upon playback of the final master at the stage set.

In order to maintain a correlation to the real stage set environment, certain guidelines and reference points had to be established. The maximum intensity of vocal dynamics had to be at the maximum level of signal hitting the tape, with all other softer dynamics appropriately lower, which was something that we constantly checked to maintain a consistency throughout the two-hour performance. It was also decided to ignore any overall level differences between vocal stacks due to location, because these could be adjusted on the stage set by turning up or down the power amplifiers supplying a particular stack.

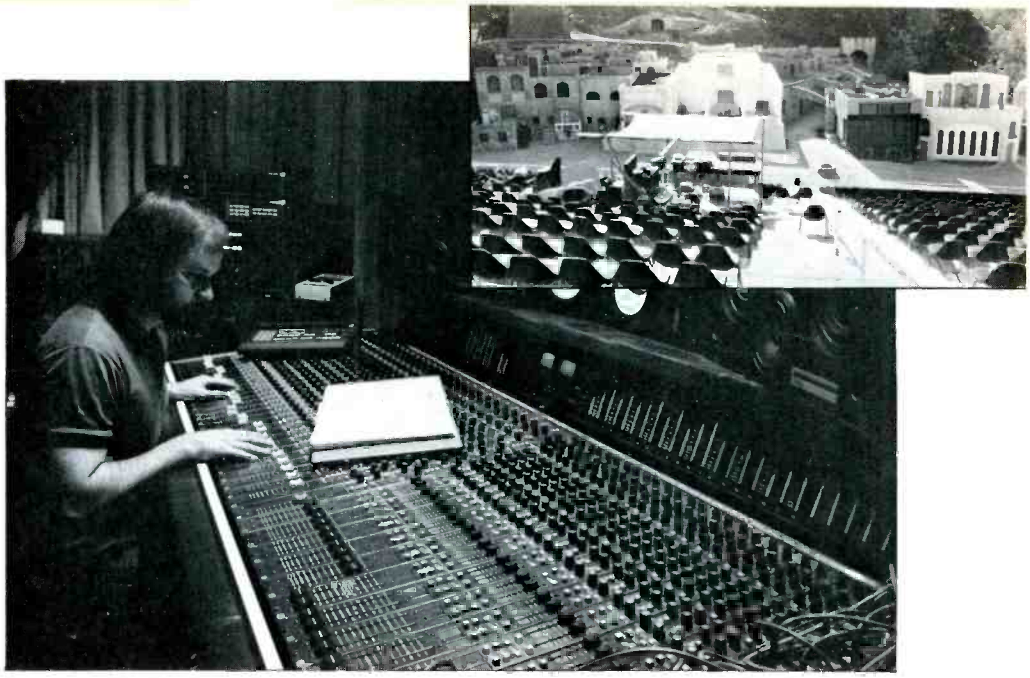
The main concern was to keep things consistent within a track. In simple cases, movements between tracks were done with panpots on the input side of the console. Where a number of the vocal tracks needed to be be panned through two or more speaker stacks, the vocal tracks were assigned to all speaker stack tracks involved, and the movements accomplished using the group faders on the output side of the console, while leaving the monitor faders stationary to ensure accuracy in the movement. Due to their excellent imaging characteristics, Westlake BBSM6-F close-field monitors were used — they proved very helpful in maintaining an accurate reference point throughout the mixing of the two-hour vocal master.

Although we tried to maintain a

proper approximation to the real environment, there were still areas which required educated guesswork. One of the speaker stacks is about 300 feet behind the others, and is intended for scenes which take place in that area, while another stack is located a considerable distance off to the left of the stage set. Because our monitoring environment isn't exactly three-dimensional, a little guesswork was involved.

One thing we learned rather quickly is that we could not pan or move the sound to these distant speakers without the audience experiencing a distracting echo because of the increased distance from the listener. In retrospect, I suppose a way of creating a 3D monitoring environment in this case would be to put the monitor for the two distant speaker tracks through an appropriate delay.

Another area that required a bit of guesswork was the levels at which to mix in the special effects and crowd noises. Having multiple stacks across the stage set was an enhancement to the effects and crowd scenes, but we had to take into consideration the overall increase in level when something was assigned to more than one speaker stack, and compensate for it accordingly.



Author Bill Cobb at Poiema Studios, combining vocal, BG and effects to a seven-track master with mixes corresponding to the speaker stacks (*top*).

Music Recording

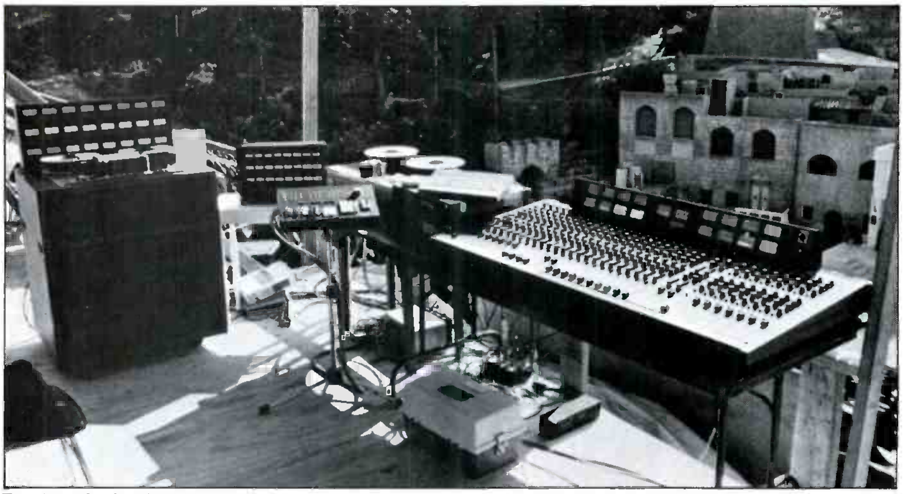
With the vocals mixed to the final performance master, the next area to be addressed was the underscoring music, written and arranged by composer/arranger Phil Perkins. We were scheduled to record the music in London the following month, but the opening date of the *Passion Play* was about a week away. The plan was to temporarily use pre-recorded music until the London tapes were ready. The pre-recorded music was mixed from a couple of sources onto the two stereo music tracks, although each side was panned slightly towards the center. The final master multitrack was then copied and shipped off to the *Passion Play* for their rehearsals and first performances.

In the meantime, we began preparing tapes for the London trip, where the music written for each scene would be recorded. For each segment of music, a two-inch tape was prepared with a mono reference mix of all of the master vocals, along with timecode on a second track. Each segment of music usually encompassed one to two scenes. A click track was then recorded on each segment to ensure that the music would be properly locked into the scene action.

We were to record a 45-piece orchestra at CBS Studios in London, during nine, three-hour sessions over a period of five days. The CBS engineer we were working with, Mike Ross-Trevor, had an excellent handle on the room and equipment and was a real pleasure to work with. The control room housed a custom-built, 56-input MCI JH-500 console and multitracks. Because of the amount of material we had to cover (two-hours total running time), we moved along at a relatively fast pace.

Combining Music and Vocal Elements

Returning with the completed tapes to Poiema Studios, we contacted



During the '84 "Passion Play" presentation, it was decided to combine the music and vocal tracks, instead of having two discrete "music stacks" on either side of the stage area. Shown here is the MCI JH-24 used to replay the original master tapes, while a Stephens multitrack recorded the new "mixed master," with music and dialog combined for each of the seven speaker stacks, and stereo spreads between adjacent pairs.

Recording Services Company to hire another MM-1200 and Shadow synchronizer. Because we had timecode and a reference track on the London music tapes, we could lock it to the master performance tapes, and scene by scene mix over the music to two tracks on the latter. Between a few of the sequences, there were cases where the music on the end of one sequence would overlap the music on the beginning of the next, due to the tightness of the cues. We fixed this by splicing in extra two-inch tape to allow the previous music sequence to end before the next started. Although this technique compromised our timecode capabilities in the edited tape, it posed no other problems since there was normally no vocal between the music sequences.

One aspect that was relayed to us from the *Passion Play* site concerned the temporary music mix. When action is taking place on one side of the stage set, and music is coming out of both sides, the music is distracting to those sitting on the side opposite the action. So in mixing the London music, we panned the mix and favored the side where the action is taking place.

Another aspect of mixing music in the control room that proved to be difficult to control was the relative level of the music to vocals. With the temporary music there would be times when the vocal would get lost under the music, and times when it should have been louder when played on the stage set. To alleviate this problem with the new tapes from London, we decided to mix the music in the studio to two tracks on the final performance master. Then I flew back out to the *Passion Play* with a portable Stephens multitrack and a small console. I set up the machine and the console in the middle of the audience area, and connected it to all the speaker stacks in the same way that the in-house 16-track is normally interfaced. I then rebalanced via the console the two orchestra tracks mixed at Poiema Studios onto two other multitrack tracks, controlling levels to complement the vocal performances. These two new music tracks were then used to drive the music stacks.

Another potential problem area that we had to deal with was ensuring that the vocal stacks are electronically and physically aligned as close as possible to one other. Because each stack is located in a different building structure, the sound can change as it is panned from one to another.

In addition to the soundtrack production and sound reinforcement improvements mentioned earlier, a new computerized lighting system has been installed. All lighting and special effects are now controlled through a Colortran Dimension 5 computer; all dimming and non-dims are also made by Colortran, and are digitally controlled. Future plans include four lasers, a total 800 kw lighting system, three follow spots, and added special effects.

Also added to the play was a complete flying system to make the ascension of Christ as lifelike as possible. A pyrotechnic system was also installed, which will be one of the most complex in the world.

* * *

All in all, the recording of the *Passion Play* was a very different and interesting experience. It opened my eyes to some of the considerations one must take into account when planning for an involved project. Everyone should do a project like this at least once! ■■■

Photography by Ralph Jones

DESIGN AND INSTALLATION OF A HIGH QUALITY SOUND SYSTEM FOR THE GOLDEN NUGGET CASINO

by Ralph Jones

Consider the assignment: a 400-seat club located in one of the finest Las Vegas casinos is to be remodeled. A new sound system is required, and it must be of the highest quality attainable; the sound to *every* seat in the house must be uniformly excellent. The specification of outboard equipment for the mix position includes most of the top-of-the-line signal processors found in major recording studios. Money is no object.

It isn't often that a consultant or contractor hears the client say: "We just want the best, and you have unlimited resources to do it." Well, the other shoe is about to drop, because there's more. In less than a month, the club is scheduled to close for extensive remodeling. The replacement sound system has not yet been designed, much less the parts ordered. Twelve days after closing, the club will re-open with Paul Anka — an absolute perfectionist when it comes to sound — headlining the bill. And the system *will* be ready for that show. Period. This job suddenly begins to look like a nightmare, doesn't it? No time for measurements, nor for speculative rambles through equipment catalogs, picking and matching components. You have one shot, and must work around extensive demolition and reconstruction. Twelve days to design and install a state-of-the-art sound reinforcement system?

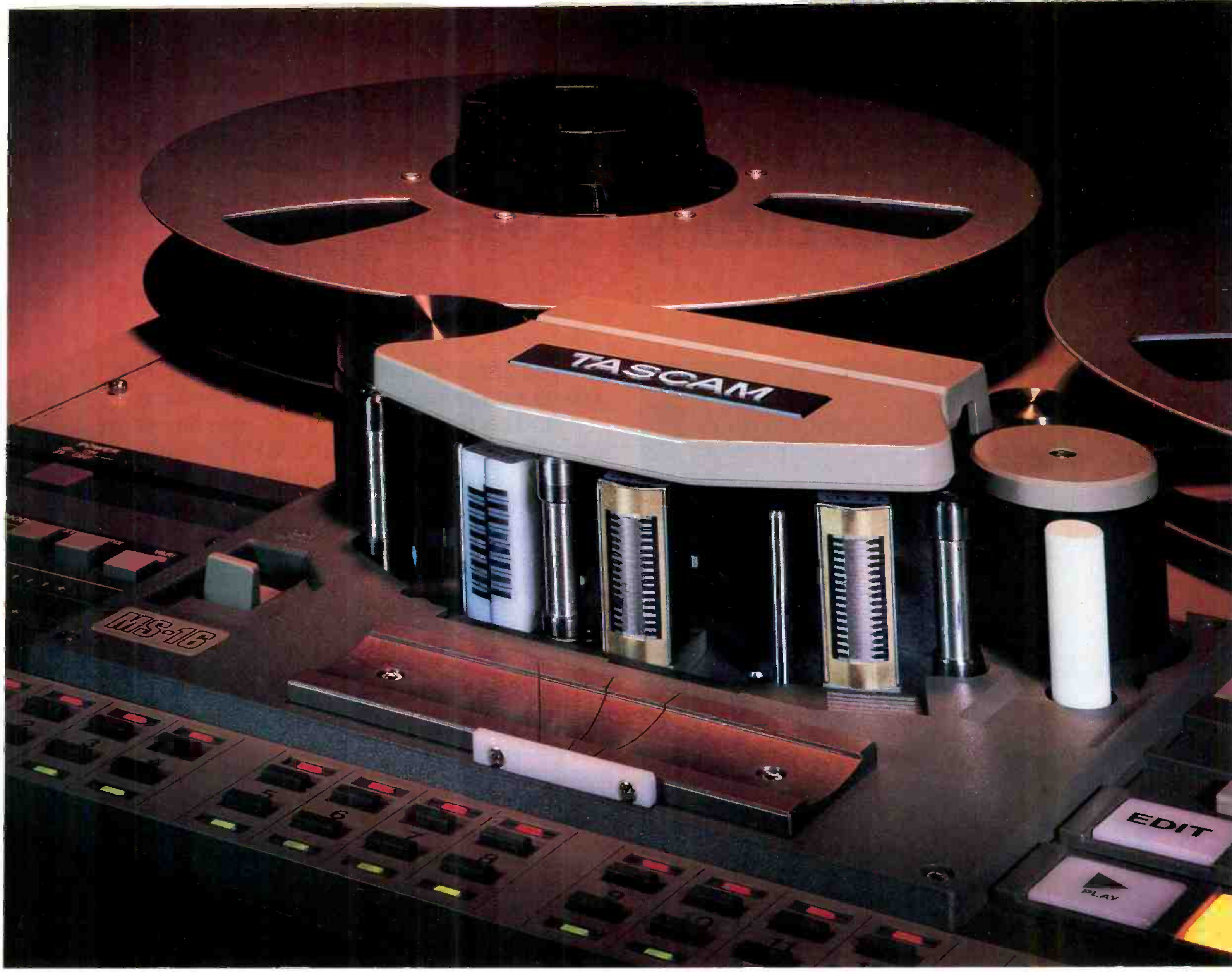
Just such a challenge was presented recently to John Meyer, president of Meyer Sound Laboratories. The story of how Meyer and theatrical sound designer Abe Jacob met the challenge is both fascinating and thought provoking.

— the Author —

Following 10 years of teaching and performing electronic music, **Ralph Jones** moved from New York to San Francisco in 1979, taking a position with Meyer Sound until 1984. Now living in Los Angeles, Jones possesses broad, professional experience pertaining to music, acoustics, electronics design, and loudspeaker technology, and currently divides his time among writing, consulting, and film scoring-assignments.

The Casino Environment

The Las Vegas Golden Nugget is a richly appointed club located not on The Strip, but in the city's downtown



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Record/Function switches for each track allow effortless, one-button punch-ins. Input Enable allows instant talkback during rewinds. With the MS-16, you're free to concentrate on the project at hand... rather than on your tape machine.

The MS-16 takes the grief out of locking up with other audio and video machines as well. The 38-pin standard SMPTE/EBU interface affords speedy, single-cable connection with most popular synchronizers and editing systems. And the MS-16's new Omega Drive transport stands up to continual shuttling while handling tape with kid-glove kindness.

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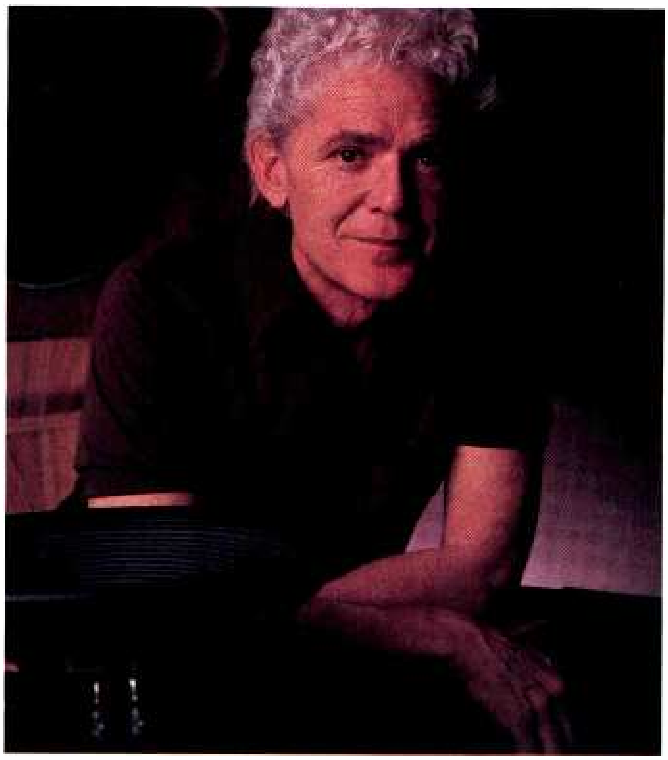
Before you choose speaker components, listen to Tom Hidley.



It's a good bet that of all the people reading this ad, 10 out of 10 know the name Tom Hidley.

One engineer we spoke with called him "the best engineer in the world." Another described him, a bit more colorfully, as "pretty damn hot."

But most of you know him as perhaps the foremost studio designer in the world today.

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In fact, he does more than prefer them. Insists Tom, "I will use only TAD, unless a client demands otherwise."

We, of course, are delighted that Tom feels so strongly. But it should also be of more than passing interest to you, since you want the speaker components you use to be the best.

And on the subject of "best," Tom has some very definite opinions about TAD. "They are the most state-of-the-art, consistent quality products today. Nothing touches their performance, honesty, stability and transient response."

"NOTHING TOUCHES THEIR PERFORMANCE, HONESTY, STABILITY AND TRANSIENT RESPONSE."

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"TAD MAKES THE BEST SOUNDING COMPONENTS I'VE EVER HEARD."

But for Tom, that's all frosting on the cake. "At the end of the day," he says, "it's what comes out of that speaker that determines success or failure. No matter what it measures, it all comes down to what it sounds like. TAD makes the best sounding components I've ever heard."

If you're in the market for professional speaker components, for yourself or a client, we hope you'll seriously consider what Tom Hidley has to say about TAD.

And thanks for listening.

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area. Owner Steve Wynn, is something of a celebrity in the town: even the cab driver who brought this writer from the Las Vegas airport knew him by reputation. Wynn is widely credited with single-handedly revitalizing the downtown area, which had been languishing because of competition from the gaudily-lit, entertainment-packed Strip.

Upon acquiring the Nugget, Wynn embarked on an extensive renovation of the aging hotel. Determined that it should become a major Las Vegas attraction, the new owner transformed the structure inside and out. By Vegas standards, the result is surprisingly tasteful. Although there is no shortage of fancy lighting or gleaming metal detail, this casino has the distinct air of class that is *de rigueur* for a club intending to attract high rollers.

Among the renovations that Wynn brought about was the Cabaret, a small nightclub that forms the focus of this present article. First opened in May 1984, the venue originally accommodated approximately 300 people, offering an entertainment atmosphere very different from the huge showrooms that are a Las Vegas staple.

"It's really a very unique room," offers Mike Slurzberg, the Nugget's head sound technician. "What he [Wynn] is doing is to bring the top level of stars into a very intimate environment. Having the audience be able to almost reach out and touch performers of this magnitude makes a performance highly personal. It's a unique experience for the performer as well — he can break out of the standard mold of huge production shows, and use just a few instruments instead of an orchestra."

Renovation Process

The Cabaret's current renovation involved significant modifications being made to the room. The sound booth was widened by three feet on each side. Above and in front of the booth, which is placed centrally in the room about halfway between the stage skirt and the rear wall, a cutout was made in the ceiling to house a follow-spot and a pair of delay speakers; to conceal the cutout, a cloud piece was added. The entire stage area was gutted, right up to the next floor above. Although the new stage is no larger than the one it replaces, it has been raised about 10 inches to make room for a monitor trough. A lighting trough was put in upstage, and a turntable added on-stage.

John Meyer (*left*) and the Golden Nugget's head sound technician, Mike Slurzberg, closely examining a graphic output plot of the frequency response for the room's main loudspeaker cluster.

When it first opened, The Cabaret was equipped with a traditional component-based system — horns and bass bins. But its new owner is an audiophile, and a quality hifi system graces his office. Accustomed to listening to playback from a CD unit while he works, Wynn was understandably dissatisfied with a merely adequate PA for his new club. Although the time allotted was brief, and the work planned already prodigious, he was determined that the renovations include a new sound system.

To demonstrate the level of quality that he wanted, Wynn played Meyer a series of recordings on his office hifi system. "What he cared about was very evident from listening to his demo," Meyer remembers. "He wanted good quality, as good as his hifi system. He said: 'Every table in my room is important'."

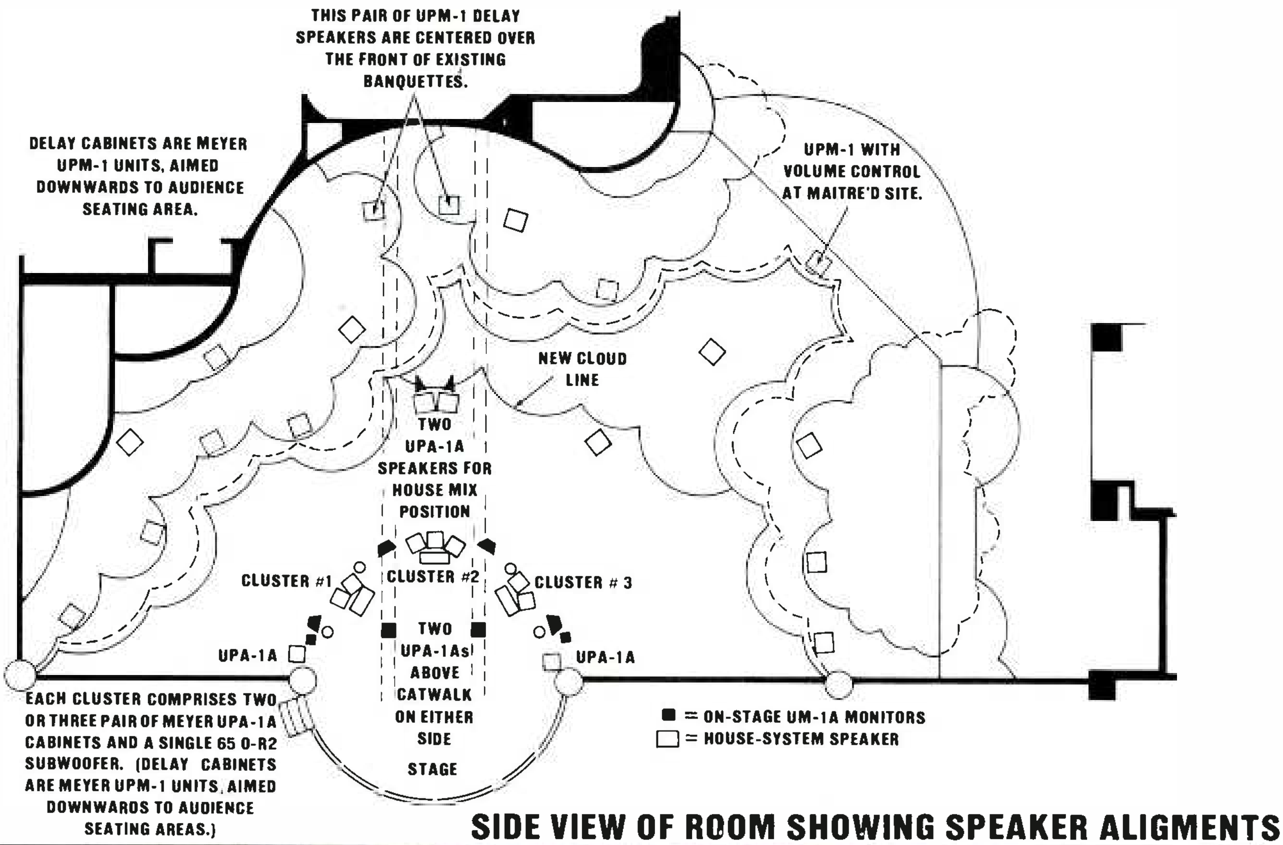
Meyer had come to meet with Wynn in response to a call from Tony Melici, the casino's stage manager. He soon discovered that the list of equipment the Nugget wanted included Meyer Sound loudspeakers. It turned out that the list had been submitted by Paul Anka's staff at the time the artist was booked to play the Cabaret, Anka has been using the company's monitors for some time.

After talking with the casino directors, Meyer suggested that they hire a sound designer for the project. "They had an horrendous list of equipment, including lots of outboard gear," he recalls. "It was clear that this system *had* to perform; it was going to be judged on the same level as Broadway shows."

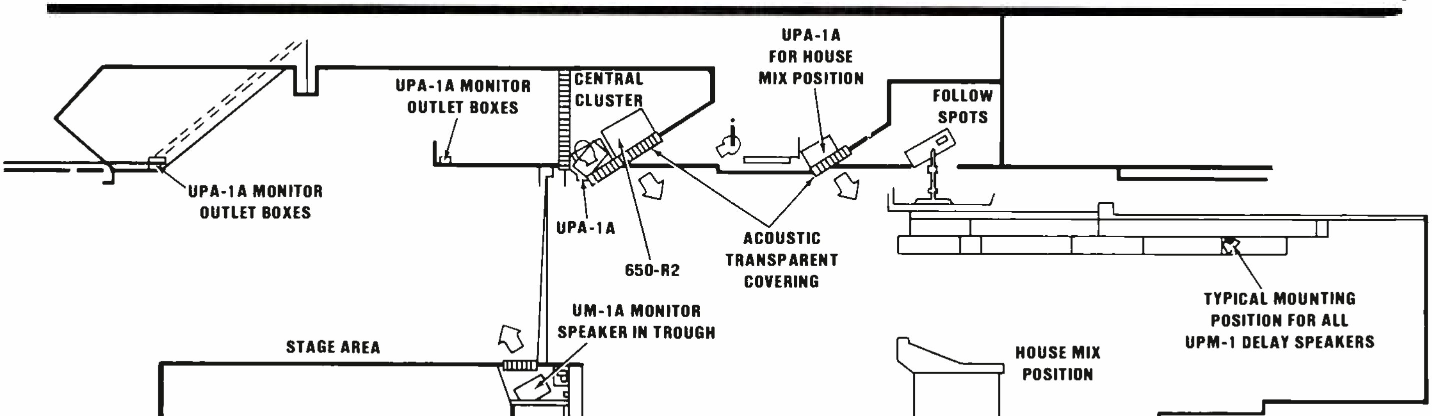
The person that Meyer suggested for the role was Abe Jacob, considered by many to be the preeminent sound designer of American theatre. Another theatre professional, Joe

Meyer Sound's head of engineering, Alexander ("Thorny") Yuill-Thornton II.

Sound designer Abe Jacob at the house mix position taking response measurements.



SIDE VIEW OF ROOM SHOWING SPEAKER ALIGMENTS



GOLDEN NUGGET CASINO FLOOR PLAN

(Note: The exact position of delay cabinets was adjusted during installation.)

Layton, had already been engaged as artistic consultant on the project, and Layton had stipulated that he didn't want any "gruesome sound problems." Since Layton had worked with Jacob on numerous occasions, and trusted his judgement, the choice was immediately agreeable.

Meyer located Joacob, and proposed the project to him. Recalling the conversation, Jacob chuckles. "He said: 'Listen, I'm doing this little room in Las Vegas.' Three hundred seats — what could it be?" The ensuing meeting in Las Vegas brought together all the principals of the project, and the race was on.

Design Phase

Given the installation's impossible schedule, the selection of an experienced theatrical sound designer to specify the system appears, in retrospect, to have been a masterful stroke. Indeed, the challenge presented by the Golden Nugget was one that Jacob faces continually in his work as sound designer for touring Broadway shows. "This was like specifying a show," he says. "It was done without the computer programming and model-making which some of our colleagues in this business deem essential. One, there wasn't the time. Two, it's the actual, practical experience of having done years of shows in these types of environments — and knowing what artists like this want — that makes this installation work. I think the results will show the value of that approach."

Relying on that practical experience, Jacob and Meyer met in New York and sketched out a specification for the system that was then finalized by Jacob. The resulting document runs to over 20 pages, comprehensively detailing all aspects of the installations, from loudspeaker placement to the power circuits and handling of grounds. "I prepared a whole list of very specific details for the installer, Ellis Electronics, to follow," Jacob relates. "We have 200 kVA of isolated power on a Topaz series voltage regulator. The outlets are hospital-grade throughout, the system. AC ground is separate from the conduit ground throughout, and we've allocated AC outlets for on-stage instrument use which are on the same circuits as the sound system.

"I also specified acoustical treat-

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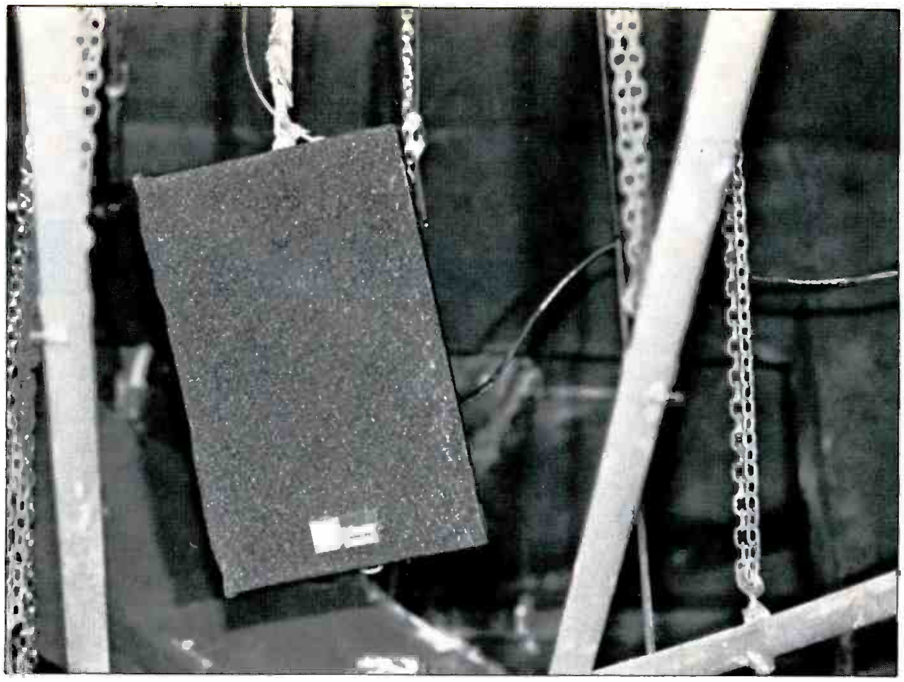
ments: glass wall insulation, and so on. All of the curtain tracks, for example, have the tin faced with fiberglass, rather than being just architecturally draped: Frank Kelley, the project designer from Fiorentino, was *very* conscious of sound."

Regarding the criteria that governed speaker placement, Jacob states, "We wanted to have a sense of uniform coverage, while retaining the impression that the sound was coming from center stage." The remark is telling: rather than rattling off a string of numbers, or citing a bewildering series of measured quantities such as Q, RT60, and so on, Jacob approaches the question by defining — in simple terms — an aural experience. Coming from a man of lesser achievement, such a statement might provoke ridicule in some quarters. But Abe Jacob is no fool: in the field of legitimate theatre, in which even the simplest reinforcement systems are enormously complex, Jacob has risen to the top. As we shall see, his remark represents an approach that was fundamentally important to the success of this installation, and which illustrates the characteristic working philosophy of these two professionals.



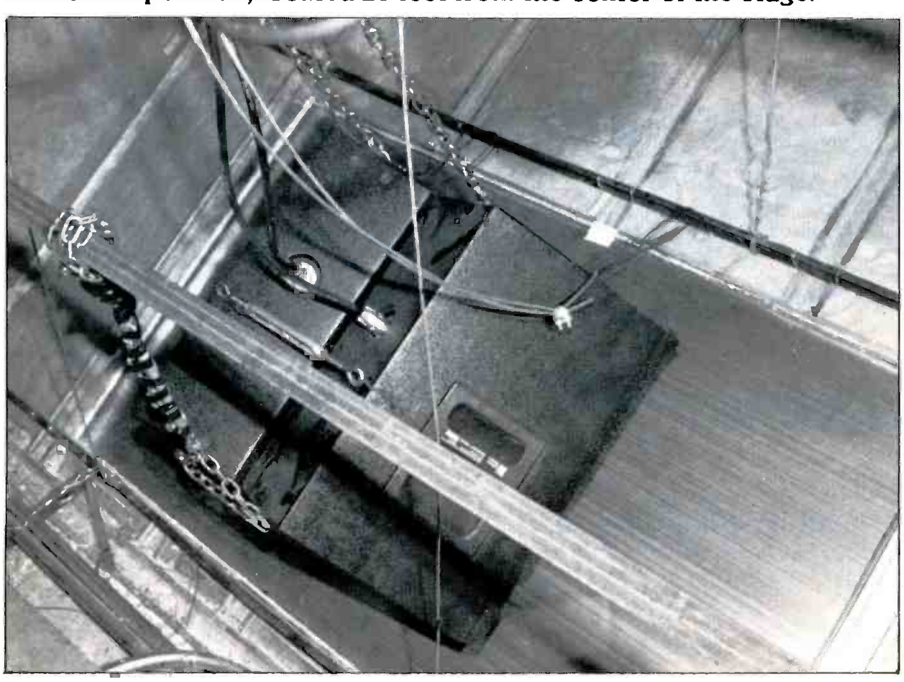
Sidefill monitors comprised a pair of Meyer UPA-1A two-way cabinets suspended above either side of the stage area from the catwalk.

Main Speaker System

As can be seen from the accompanying diagrams, the Cabaret is a roughly rectangular room, with the stage placed against one of the long walls, dividing the room into two unequal areas. The stage itself is semicircular, with the consequence that the required coverage angle for the main system is 180 degrees.

In order to provide even coverage to the room, and deal with unequal coverage areas on either side of the stage, Jacob and Meyer specified three main speaker clusters located on the periphery of the lighting grid above the stage. Each main speaker cluster employs UPA-1A cabinets. Designed to be a full-range arrayable component for moderate-scale sound reinforcement, the UPA-1A is a two-way system with a 12-inch low-frequency driver and 40- by 80-degree high-frequency horn, and requires an integral signal-processing package. The M-1 Control Electronics Unit performs the functions of electronic crossover, factory-preset equaliztion, time correction, and Meyer's SpeakerSense™ driver protection limiting.

An additional UPA-1A cabinet on a delay provided monitoring at the house-mix position, located 25 feet from the center of the stage.



The central main cluster comprises an array of three UPA-1A cabinets, while the left and right clusters each consist of two UPA-1As. Within each cluster is allocated a separate M-1 electronics unit, in order that each may be controlled independently for purposes of balancing the system (more on this later). Since the frequency response of the UPA-A1 system alone drops off rapidly below about 60 Hz, each main cluster is augmented by a Meyer 650-R2 subwoofer. This system consists of two 18-inch drivers mounted in a single vented enclosure; like the UPA-1A, the 650-R2 requires its own B-2A processing package. The specified frequency response of the combined system extends from 20 Hz to 16 kHz.

With relatively powerful subwoofers installed in the catwalk, some attendant acoustical treatment was necessary. "The catwalk area is a nightmare of open tin surfaces and metal-to-metal contacts," Slurzberg explains. "We had to isolate the upper area and surface it in one-inch fiberglass, then went around chasing down vibrations and damping them. We have extra fiberglass and damping materials in storage, in case we find later that improvements are needed."

The mix position in the Cabaret is a sunken area located in front of center stage, about 25 feet away from the

. . . *continued on page 114*

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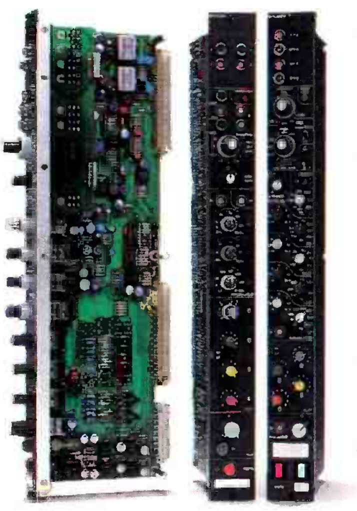

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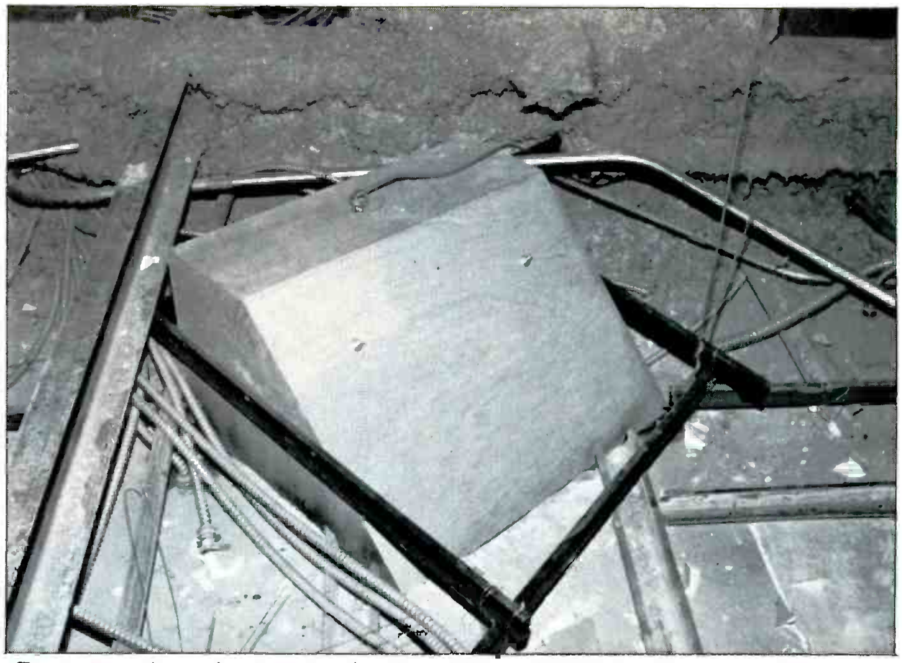
GOLDEN NUGGET SOUND SYSTEM

— continued from page 110 . . .

stage lip. In order to provide a balanced reference signal for the house engineer, a pair of UPA-1As on a delay are mounted in the follow-spot ceiling cutout just in front of the mix position. Again, for purposes of balancing and equalization, these speakers share a separate M-1 electronics unit.

Jacob and Meyer also specified a number of additional delay speakers to fill distant seating areas. At all of these positions, UPM-1 units were used. The UPM-1 is a very compact system designed specifically for delay-fill applications, and consists of a vented cabinet housing two, five-inch cone drivers and a horn-loaded piezoelectric tweeter element; its specified frequency response in free air is 150 Hz to 20 kHz. The system employs a rather complex passive crossover that might be termed "quasi-three-way": at the lowest frequencies both five-inch drivers are active; in the midrange region, the lower of these is crossed out, to avoid path-length cancellations. (The piezo element, of course, handles the high-frequency band.) The result is described as offering a very smooth polar response on both axes, and a nearly symmetrical 60- by 80-degree dispersion pattern. (Like all other Meyer products, the UPM-1 requires a processing unit, designated the P-1.)



Rear view of wooden housing for the additional UPM-1 cabinets used as delay speakers throughout the audience seating area. A total of 15 delay cabinets and associated P-1 processing units were supplied for the Golden Nugget Casino, and set up on five discrete amplification channels.

The delay UPM-1s are used to fill seating areas shadowed by the cloud shapes that jut out of the club's ceiling. A total of 15 delay UPM-1s — one for each table in the delay areas — is installed behind grills in the ceiling. Five delayed channels are available: the mix position; stage left; center; stage right; and over the maitre d' hotel's podium at the club entrance. This last pair was installed so that patrons could hear full-range sound upon first entering the club; its input signal has been routed through a pad installed in the podium, so that the maitre d' can turn it down when talking to patrons. Klark-Teknik DN700 delay lines are used throughout, and speakers on common delays are all positioned within +1 foot relative to the main cluster.

Stage Monitoring

The new Cabaret stage is provided with a trough around the periphery in which stage monitor speakers are concealed. Designated the UM-1A, the Meyer monitor used is a kissing cousin of the UPA-1A: it employs the same driver components, has the same specified frequency response, and requires the same processor. In contrast to the UPA-1A, however, the UM-1A — also known as the UltraMonitor ™ — is fitted with a 60-degree symmetrical-pattern high horn so that HF energy on-stage may be confined to the desired coverage area. The UltraMonitor cabinet also differs from that of the UPA-1A, being of a low-profile, stant-monitor configuration.

A total of eight UM-1As are available in the trough, and wooden cover sections have been made to block out

Front view of UPM-1 "quasi" three-way cabinets utilized for delay speakers at 15 locations within the audience seating area.

unused areas of the trough for smaller shows. Two UPA-1As, suspended from the catwalk above and on each side of the stage, are used for monitor side-fill. Each individual monitor channel has its own dedicated M-1 processor, thereby providing monitor sends for even the most complicated act.

One key to making a show successful in a small space like the Cabaret is to keep stage levels — particularly monitors — under control. "All the headaches you would have with stage level in a larger room are amplified in a room this size," Slurzberg offers. "You need to keep stage level and stage bleed into the house to a minimum, isolating the stage from the house as much as possible." Slurzberg is counting on the UM-1A's controlled pattern to help him do this. For cases in which that system cannot be used, however, he has retained additional monitor sends, and plans to experiment with headphones for stage monitoring.

House Mix Position

"This room has to serve a number of different acts with varying musical styles," Jacob observes. "We took that into account in the choice of processing equipment and its layout. Some of that equipment may never be used until the artist who needs it comes in, but there is the capability to do any kind of special effects that an artist would require."

The house console is a 32-input Yamaha PM-2000. Racks located underneath and to either side of the mixer hold an impressive array of outboard signal processors, including dbx 900 Series rack loaded with two Model 903 "Over Easy" compressor-limiters, three 904 PLM™ noise gates, and two 905 parametrics. These units are complemented by two dbx Model 165A and three Model 160X compressor-limiters, an Eventide Harmonizer, and three EXR-IV two-channel Exciters. In the reverb department, Lexicon's line is well prepresented: a Model 200, Prime Time II, Super Prime Time, and two PCM-42 delay processors keep company with a 224X, plus a Yamaha R1000 digital reverb. Three Yamaha six-input Series M406 stereo mixers share rack space, and a Tascam Model 122 cassette deck is available for tape playback.

All of the mix and processing equipment is installed in a sunken area at the center of the club. Since there is no door in the low wall surrounding the mixer, nor any stairs into the mix booth, once he's installed at his post the engineer is there to stay until closing. (At least he can communicate with his colleagues over a Clear-Com intercom system.) Signals from the stage reach the mix position via mike lines and through an HME additive-diversity wireless system. A Bruel & Kjaer 2230 SPL meter also is availa-

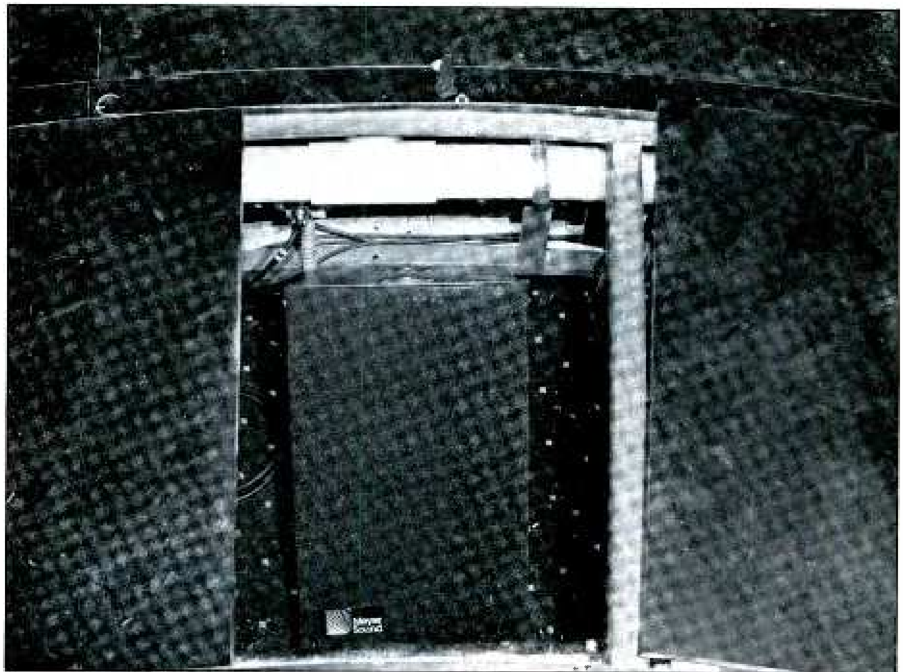
. . . continued overleaf —

ble for checking sound-pressure levels.

System Signal Routing

Access to the house speaker system is through a patch panel located at the mix position. The normalled input is mono, and feeds the entire system simultaneously, although patch points have been established to allow flexibly determined feeds. At the maximum, there are three separate main cluster feeds (left, right, and center); a separate feed for the center (mix position) delay speakers; and a feed to the delay UPMs.

Given that seating in the Cabaret is pretty much evenly distributed around the stage, one might wonder where the concept of left-, center-, and right-channel feeds came from. According to Jacob, "It was always intended to be a normalled mono feed from the console. However, one of the requirements from Paul Anka's sound people was the ability to feed a different mix to each side. He's got horn players who sit on one side of the stage, and they're a pretty powerful acoustical source. He asked to be able to feed more of their signal to the opposite speakers, so that it would be more evenly balanced."



A total of eight UM-1A UltraMonitors were mounted in a specially constructed monitor trough around the periphery of the circular performance stage area, beneath an acoustically transparent grill and mesh cover.

Clearly, there is an unusually high degree of control built into the system, which should provide flexibility in fine-tuning the distribution of sound in the space. One suspects, nonetheless, that the majority of mixing engineers will be happy with the single main system input.

The various loudspeakers are powered by Crest Model 4000 amplifiers throughout. The units are located in the ceiling of the catwalk (main PA rack and delay rack), and in a booth on the ground floor (monitor rack). Accompanying them in the racks are the respective Meyer control units for the various loudspeakers, Klark-Teknik delay units, and system equalizers. Each individual amp runs off of its own 20-amp circuit, providing plenty of AC power to back up the system headroom.

Installation Process

"Implemented poorly, a job like this could be awful," Meyer observes. "The biggest point that I made in our meetings was that the difference between sounding good and sounding bad is a *very* fine line. It doesn't take much to destroy something like this: the work has to be done with the utmost care just to get satisfactory results, and to get excellent results requires absolute attention to detail. Every bit of wiring has to be right; every phase has to be correct."

The designers determined that to meet these demanding criteria, and to cope with any unforeseen eventualities that might arise during construction, the project required on-site supervision in the construction phase. Unfortunately, Abe Jacob was otherwise engaged. "I brought in Peter Erskine of Theatre Technology in New York as my project supervisor," he explains, "because at the same time that all this was going on, I was in Barcelona doing *Chorus Line.* Peter did all the on-site supervision for the two weeks until I arrived from Spain."

"Installation of the audio system was very smooth," Slurzberg relates. "We have an outside construction contractor, Sierra Construction, who does everything for the hotel. This job couldn't have been done without their responsiveness and flexibility. The system was being designed during construction, and changes were made in the plans as the work proceeded. Sierra was always responsive to these changes."

Impressed with Sierra's adaptability, Jacob offers the following story: "When I came one day and looked at the speaker positioning, it appeared that, because of the window walls on either side of the stage, attempting to make the side speakers fill the entire side areas would put too much energy on to the windows, and cause some potentially harmful reflections. I wanted to add another pair of UPAs to cover the row of seats closest to those windows. At 10 o'clock in morning I told the contractor supervisor. He told the electricians, and by 2 o'clock the conduit had been run. By 4 o'clock the wires had been run through the conduit back to the amp position. By 8 p.m., the unit strut had been welded in to suspend the speakers."

"I wouldn't advocate doing something like this *so* fast," Meyer avers, "but having that flexibility, and the designers on-site, avoids the problems of ending up with air-conditioning ducts in front of speakers and so on, which occur all the time because of changes in construction planning."

The sound-system installer was Ellis Electronics, a local contractor whose work normally runs to surveillance cameras, alarm systems, and so on. According to the designers, Ellis provided valuable expertise in wiring methods and codes, and made sure that there was plenty of transformer-isolated power for the system.

Even Meyer Sound was called upon to respond to last-minute needs, as Jacob relates: "We found that it was not suitable to hang the delay UPM-1s and disguise them within the palms, as we had originally planned. We decided to mount them in the ceiling above with grills, so Thorny [aka Alexander Yuill-Thornton II, Meyer's head of engineering] went back and,

MEASURING LOUDSPEAKER FREQUENCY AND PHASE RESPONSE WITH PROGRAM MATERIAL WHILE AN AUDIENCE IS PRESENT

The measurement and equalization of loudspeaker systems *in situ* is a long-standing practice in professional audio, and with good reason. A loudspeaker system in an enclosed space will always measure differently than it does in free field, due in large part to resonances within the space. Even if the speaker in question exhibits a flat frequency response in free field, the combined response of the loudspeaker and room will deviate from flat.

In the sound reinforcement field, a common practice has been to measure and equalize a system in the empty hall, since existing measurement techniques have required, without exception, a specific excitation signal; test signals are not only notoriously unmusical, but downright irritating to an audience. Yet it has been known for some time that the presence of an audience can affect rather drastically a room's resonant characteristics, particularly if that room is an arena, convention hall, or reverberant church. This being the case, measurements of a system made in an empty hall do not provide an accurate gauge of its performance when the hall is full, least of all in the sort of acoustical conditions that sound reinforcement professionals most often face.

Source Independent Measurement (SIM), developed by John Meyer, is a special case of dual-channel fast Fourier Transfer (FFT) analysis. The crucial aspect of SIM, as its name suggest, is that it does *not* require a specified excitation signal in order to derive the frequency and phase response of a loudspeaker system in a hall. In fact, SIM does place some broad constraint on the excitation signal if it is to achieve reliable results, but these are quite practical. The most important constraints is that the signal excite the full frequency range of interest in a substantially random manner. In other words, the excitation signal could be simply the one for which the system is used, since music and speech can satisfy both requirements. For this reason, SIM enables one to measure and fine-tune a concert reinforcement system during the concert — or, for that matter, a church system during a service — using the program itself as the test signal, and thereby optimize the system's performance for actual operating conditions.

Measurement System

Shown overleaf is a diagram of the basic measurement system used for SIM. The equipment required is a calibrated microphone and preamp; a high-quality adjustable delay line (a well-implemented CCD bucket brigade is sufficient); and a two-channel FFT

GOLDEN NUGGET SOUND SYSTEM

in a day, designed enclosures which Sierra Construction then fabricated from drawings he shipped down to us. I found, by the way, that I didn't have to reposition any of the UPMs after I arrived."

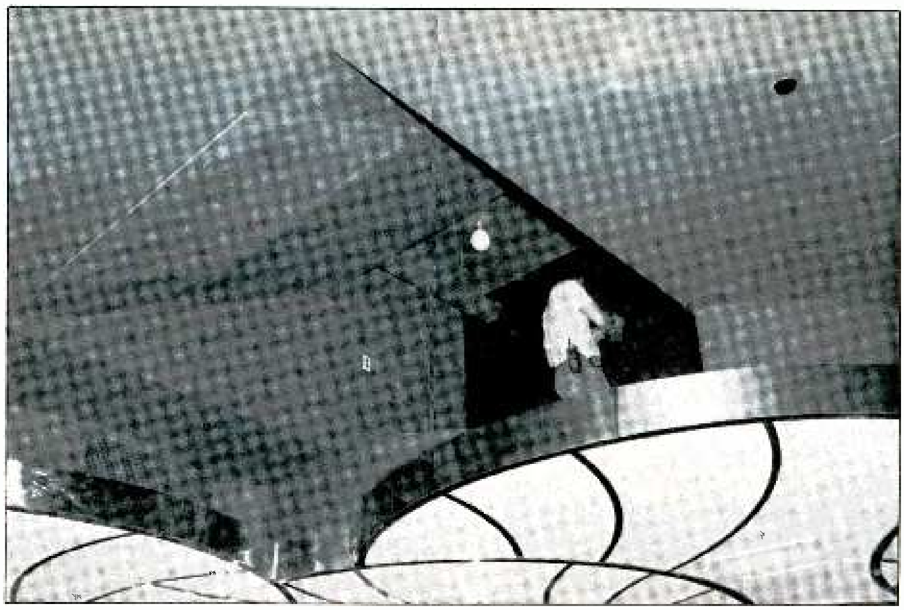
Balancing and Equalizing the System

"I knew from experience," Meyer states, "that we were going to need EQ. These systems add together very well but, even if everything is flat to begin with, once you get into the room the sound's no longer flat: some frequencies have made it better than others. EQ becomes more important the deeper you go into the room, away from the main system."

For the purpose of room equalization, 12 Meyer CP-10 two-channel Complementary Phase Parametric Equalizers were included. The CP-10, introduced at the October 1984 AES Convention, provides five sections of parametric EQ per channel, plus high and low-cut shelving controls. The EQ sections reside on separate, field-replaceable cards, each of which is furnished with its own in/out switch. Individual filters are of the reciprocal type: the cut curve is a mirror-image of the boost curve. (This is not normally the case with parametrics, even though reciprocal filters are considered to be by far the best choice for loudspeaker equalization.) The CP-10's panel markings are also calibrated, and the ganged potentiometers used in the unit are said to be computer-tested and selected such that the tolerance of a given setting is the width of the calibration marks. The unit's circuitry, developed by Meyer, is described being capable of generating the *exact* complement of a resonance in both amplitude and phase, canceling the resonance while simultaneously correcting, rather than further degrading, the phase response.

At this writing, and for the purposes of the system's initial balancing, one CP-10 is assigned to each of the three main system clusters; one to each of the five delay channels; and four reserved for monitor equalization. Meyer performed the initial equalization and balancing of the system in a single day, employing a Hewlet-Packard portable spectrum analyzer and Bruel & Kjaer microphone to gather system response data.

Adjustments to the system began with the central cluster, and the subwoofers. Having balanced the subwoofers with the UPA-1As, making certain that they were adding correctly in the crossover region (around 100 Hz,) Meyer used the CP-10 to smooth the response as measured by B&K microphone mounted on a stand beneath and slightly in front of the cluster. The resulting response curve was stored in the memory of the HP analyzer, for use a reference when balancing the delay units. Meyer then measured and equalized the left and right main clusters separately, setting them as well for a flat energy response so that they could be run independently. The gain of the left and right clusters, however, was dropped slightly below that of the main cluster, so that when all three ran together the main system appeared to



Close-up detail of the ceiling cutout provided for a follow spot, and UPA-1A monitor cabinet provided on a delay channel for the house-mix position.

MEASURING LOUDSPEAKER RESPONSE — continued . . .

analyzer (Meyer has used the Hewlett-Packard 3582A, and the new Bruel & Kjaer Model 2032 Dual-Channel Signal Analyzer).

The analyzer operates in the "transfer function" mode, in which it compares two correlated signals, which it normally are the input and output of a network. The input signal to the network is taken as the measurement reference, and compared to the output; the calculated relationship between the two is a function that describes the network's dynamic behavior — in the case of SIM measurements, the frequency response. Since the method is based on a comparison of output against input, any variations in the input signal are eliminated in the computation.

In the SIM measurement procedure, the network under test is a loudspeaker system *and* the space in which is located. The excitation signal (taken from the console output) is split both to the system under test and to the analyzer reference input. A delay line is inserted in series with the reference signal input to compensate for the propagation time delay corresponding to the distance between the loudspeaker and microphone. The preamplified signal from an instrumentation microphone is connected to the measurement channel input of the analyzer. Since the music or voice signal is an approximation of a random input, RMS averaging is used.

Taking Measurements Using SIM

Meyer's procedure, as described in the main article, is to equalize the system for flat response in the near field prior to taking SIM measurements. Since only near-field measurements are involved at this point, this may be done in the empty hall using a pseudo-random noise or impulse signal, and single-channel FFT. (This step minimizes the variable of the loudspeaker itself, so that the measurements taken in the presence of the audience will deal predominantly with the room effect).

Once the system is set flat in near field, the next step is to prepare the test setup for the SIM measurement procedure. With the equipment connected as above, and the measurement microphone placed in the desired location, the reference channel delay is fine-tuned before the audience arrives, using pseudo-random noise or a repeated impulse. When the delay has been adjusted, a cassette tape of music, played as the audience enters, can serve as the test signal for initial measurements.

Meyer recommends that the hall be at least half-full before any equalization is attempted, but cautions that — depending on many factors, including hall acoustics, directional characteristics of the loudspeakers, and the ultimate audience size — the system's measured response will continue to change as more people enter.

the ear to be the dominant source.

Having optimized the main system as a whole, and stored its response as a reference, Meyer proceeded to balance and equalize the delay systems one at a time. The measurement microphone was placed in a typical listening position within the field of each delay system. With the main system running, the level of the delay system was then brought up slowly until it just added with the main system response. Meyer then noted how the delay filled in areas where the main system's response had fallen off. This left a composite curve that was matched to the stored main system response through equalization of the delay system.

It should be noted that this procedure equalizes the delay speakers for best addition with the main clusters, but without adding energy to the overall system. The effect is that the center cluster appears to the ear to be the sound source. The front-to-back sound pressure differential in the room was held to about 6 dB; without the delay systems, at the back of the room, some frequencies fell off by as much as 20 dB. As a result, the delay speakers were filling in areas of the composite frequency response that were weak.

The aural effect to this writer's ears was quite impressive. Seated in the field of any of the delay systems, one had the distinct impression of clear, full-range sound emanating directly from the stage area. The delay speakers were not audible as separate sound sources in the room; in fact, their effect was only noticeable if they were turned off. When the entire system was running, one would never have guessed that delay speakers were being used at all, which represents a distinct and pleasing change from the annoying and aurally confusing impression that delay systems sometimes can give when not properly implemented.

Source Independent Measurement Technique

This is the point where the textbooks say to stop, and in every sound system installation, no matter how extravagant, tuning of the system in an empty space represents the final step. In common practice, the realm of taste and intuition lies beyond that point, and any equalization in the presence of an audience is done by the mixing engineer. But the technical sessions of last October's AES Convention saw the debut of a new measurement technique originated by John Meyer, and dubbed Source Independent Measurement (SIM™). By employing a dual-channel FFT analyzer in the transfer-function mode,

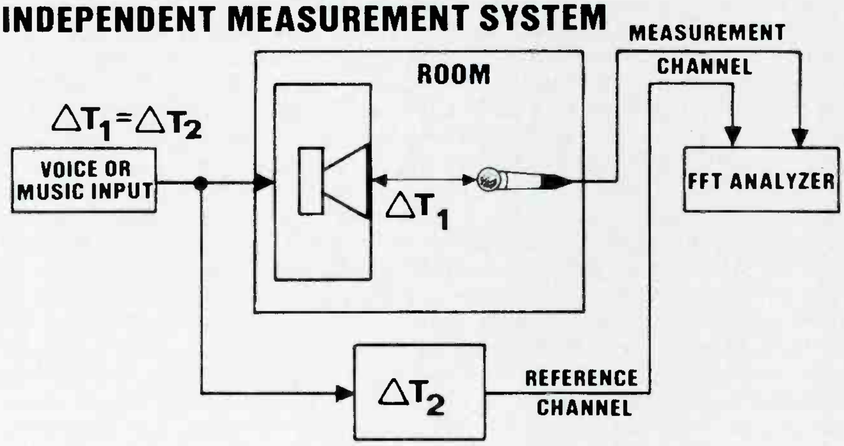
MEASURING LOUDSPEAKER RESPONSE — continued . . .

Successive measurements taken as the hall fills will provide a picture of those changes, and from these data predictions may be made about the necessary equalization. If some judicious equalization is done beforehand, the final settings dictated by the SIM measurements will not necessarily represent radical changes. It is advantageous to attempt some equalization before the event begins, if possible, in order to avoid making obtrusive equalization changes during the event.

Use of FFT in Acoustic Measurements

Some limitations apply in using the SIM technique, one of the most important being that the operator should have a good understanding of the Fourier transform and its implementation in the particular analyzer used. For example, the HP 3582A referred to above uses a linear transform (as distinguished from a "constant-Q" transform) with 125-line resolution, and produces a linear-frequency display. The linear display alone requires some getting used to if you are accustomed as everyone in audio is to a log-frequency scale. But the

BLOCK DIAGRAM OF SOURCE INDEPENDENT MEASUREMENT SYSTEM



MEASURING LOUDSPEAKER RESPONSE — continued . . .

resolution limitation also necessitates using seven different capture-time ranges to obtain adequate resolution over the audio frequencies of interest. (The B&K 2032 avoids these problems by implementing the transform with 800-line resolution. The data may be displayed on a log-frequency scale, and the entire audio-frequency range may be analyzed with very high resolution using just one range change.)

Loudspeaker systems with poor time response may be difficult to measure accurately using an FFT since the inherent delays in the speaker can cause data to fall outside the record length (time window). For this reason, the time response of the system needs to be well-controlled, and the operator must know its limits to be assured confidence in his data. If reflections up to 100 msec. are to be included in the measurement, then the time window must be at least 100 msec. long. Loudspeaker systems that are highly non-linear also may be more difficult to measure (although non-linearities will pose problems no matter what measurement method is used). Contamination of the data from crowd noise or other ambient interference must be considered, and any contaminated data rejected. (Meyer's AES paper, preprint #2150, details a curve-smoothing technique for dealing easily with this.)

Such considerations do not necessarily invalidate the technique; in fact, since SIM is based on the dual-channel FFT analysis using a random input, it will provide the best linear fit of any technique when systems with a non-linear component are tested. SIM, like any measurement method, must be employed with care and an understanding of the process. It should be noted, moreover, that the SIM procedure is intended to be used for measurement of audio systems that exhibit reasonably low harmonic distortion and controlled phase response. But this restriction seems entirely consistent, since the purpose of the technique is to allow a higher degree of refinement of a system's performance than has been possible heretofore. Loudspeakers with substantial distortion are difficult to measure no matter what method is used; they do not respond reliably to equalization; and they sound terrible. For such systems — and for systems with poor phase response — equalization in an empty hall is sufficient, since changes in room resonance due to an audience will not be the dominant factor contributing to degradation of the sound quality.

Since a comprehensive guide to SIM is beyond the scope of this present article, readers who are interested in employing SIM are referred to John Meyer's paper, "Equalization Using Voice And Music As The Source," AES preprint #2150 (I-8). Look also for further coverage of this and other dual-channel FFT techniques in a future issue of *R-e/p*. □□□

GOLDEN NUGGET SOUND SYSTEM

SIM is said to enable the user to compare the input signal to a loudspeaker system with that of a sampling microphone placed in the space. By thus deriving the transfer function of the overall room/loudspeaker system, SIM effectively eliminates the need for artificial test signals, using instead the program material itself as a reference. The truly revolutionary implication of this technique is that it enables — for the first time — non-intrusive measurement and adjustment of a reinforcement system in the presence of an audience. (Further details of the analysis technique are included in an accompanying sidebar.)

"Prior to John delivering the AES paper," Abe Jacob recalls, "I had seen an advance copy, and we had a conversation about the possibility of using his equalization procedure in the theatrical world. There are obvious and exciting advantages to being able to equalize a system in the presence of an audience during performance. When we became involved with this installation, it was only natural that the SIM process be included in the overall plan."

"One of the main reasons I wanted to do the project," Meyer relates, "was to have a first-hand chance to try the process in a club like this. The Cabaret system is obviously not a typical club PA — most people think of a club PA as a couple of speakers in a cluster. We're showing that, in order to get the kind of performance that artists like those playing here demand, you can't do that anymore. If you want to walk around the room and have it sound spectacular everywhere, you *must* deal with the effect of the audience in the space."

"We talked about doing an initial study," the designer continues, "and this is it. When we open with the first shows, there's going to be a fair amount of sound absorption, given the number of people that will attend performances in a room that's about 10,000 cubic feet. We want to see how much it's changed, then make adjustments. I feel we're already better off than you generally would be after a lot of work. But the most important work will come when the room fills up with people."

To effect final adjustments based on SIM data, Meyer and Jacob specified another CP-10 equalizer, patched at the console output. "Theoretically," says Meyer, "the room shouldn't change that much from half-full to full, but from the study we've been doing lately, that doesn't seem to be true. People seem to keep absorbing. This room has seats, and rugs on the

An equipment rack houses the P-1 processors for the UPM-1 delay speakers, and CP-10 complimentary phase equalizers

floor, and the changes may be less extreme than you see ina large concrete hall, but I expect some weighting of the overall balance. We can handle this from the output of the mix console. Since the room as a whole is equalized fairly flat right now, it will track what we do.

"The important thing that we've done is to be sure that the sound is even and flat everywhere, and tracks the input. Then, in concert, the overall balance can be handled at the mix position."

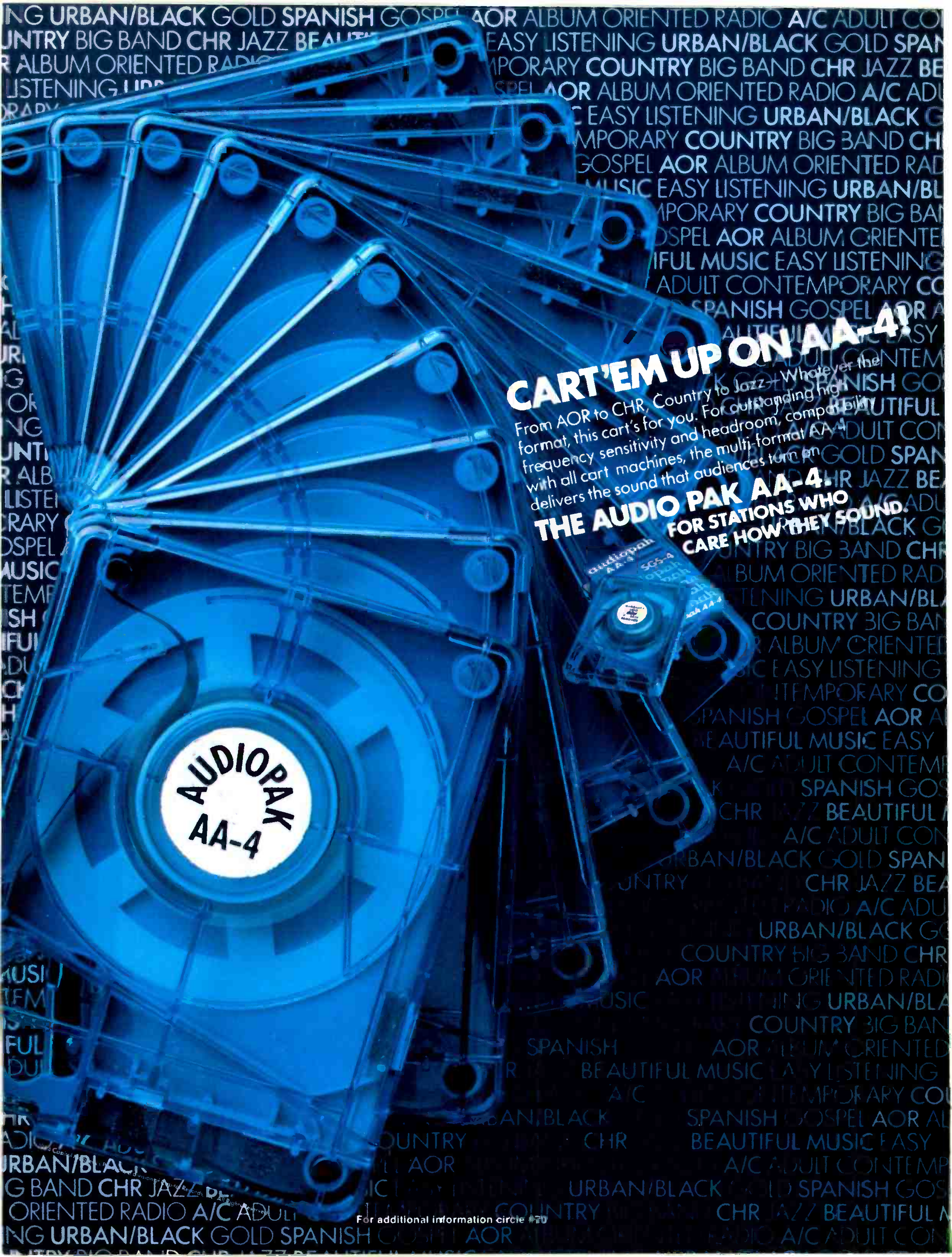
Meyer and Jacob used the SIM technique to measure and fine-tune the Cabaret system during opening performances by Paul Anka, the first artist to use the new sound system. Measurements were taken using a Hewlett-Packard 3582A duel-channel FFT analyzer in conjunction with the same B&K sampling microphone that was used in the initial equalization procedure. The output of the mix console, sampled just before the CP-10 equalizer in the signal chain, was taken as the reference. Tapes played before hand, and Anka's performance itself, served as the test signal sources, the CP-10 equalizer being adjusted for the best match betweeen the console output and the sampled room sound.

While subjective impressions are the most difficult to convey and verify, it must be said that the effect of final equalization on the Cabaret system is impressive. By minimizing the final variable of audience absorptions, and the consequent response anomalies, Jacob and Meyer have produced what must surely be one of the highest-fidelity reinforcement system in existence. Such an achievement surely is easier under relatively controlled the conditions of a permanent installation than it would be on the road. But, especially given the job's schedule, the Cabaret installation may well prove to be a landmark in the field of sound reinforcement. Moreover, it seems likely that increased application of SIM in theatre and concert sound might very well elevate the technical level of the concert-sound industry as a whole, to the benefit of sound engineers, performers, and audiences alike.

Assessing the Project

Over dinner in the Golden Nugget's restaurant, Abe Jacob, John Meyer and Mike Slurzberg discussed their feelings at the end of the grueling period of installation and fine-tuning of the Cabaret system. "I'm sure many of our colleagues will take umbrage atthe way in which this system was designed and installed," Jacob offers, "but it is the *final* result that is of primary importance." Slurzberg, who is charged with the responsibility of maintaining and operating the system, echoes Jacob's sentiment: "In the final analysis, what counts is what you hear when you come in and sit through a performance."

"The experience that I've had with major installations in the past," he continues, "is that they come from someone who has been too close to their drawing board for far too long. The opportunity to have this kind of talent come in and help to implement what I need to do my job — people who have experience both on the drawing board and in live performance, and know what it takes to do a show — is very rare. It's made a really difficult job possible for me. All along the line, everyone has been very receptive to my needs as the operator of the system." ■■■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FACILITIES EQUIPMENT PEOPLE

North-East:

□ **THE RECORD PLANT** (New York City) has added a 56-input **GML Moving Fader Automation System** to its **Trident TSM** console. According to general manager **Mitch Plotkin**, "In 22 years in this business, I've never been so extremely impressed by a piece of equipment or the staff of any organization as I am with George Massenburg, his staff and this automation system. My entire staff is in awe and we are one picky group. This system alone will reinstate the Record Plant as a major force on the East Coast." The recent sale marks the eighth GML system to be installed; it utilizes two Motorola 68000 microprocessors in conjunction with five other processors, 500 Kbyte of RAM and a 20 megabyte hard disk to control servo faders. The next systems are scheduled to to be installed at Record One and Bill Schnee's Los Angeles Studio. *New York, NY.*

□**NIBOR RECORDING STUDIO** (Hurley, New York) has added an **E-mu Systems Emulator 2, Apple Macintosh Computer, Yamaha DX7, LinnDrum, Lexicon** digital reverb, and a **Sony PCM-501ES** digital processor. *Hurley Mountain Road, Hurley, NY 12443. (914) 331-3060.*

□ **MEDIASOUND** (New York City) has added a **Studer A80** four-track and specially designed drum platforms to Studio A, plus an additional **Studer A800 MKIII** 24-track with **Dolby PSI** NR rack, **Studer A80** four-track, **Audio Kinetics Q.Lock 3.10** and **Adams-Smith** timecode systems to Studio B. Studio C now boasts a new **Harrison Raven** 32-input console and **UREI 813B** loudspeakers driven by an **H&H M900** amp. *311 West 57th St., New York, NY 10019. (212) 765-4700.*

□ **SKYLINE STUDIOS** (New York City) has added a custom-designed **Solid State Logic SL 4000E** console. In addition, the studio has formed a collaboration with musician **Peter Scherer**, the owner/operator of the new on-premises **New England Digital Synclavier II**. The new console has a 56-input mainframe fitted with 40 channels, **Studio Computer**, and **Total Recall**. Built into the producer's desk are vintage **Neve** and **API** equalizers. Eight miles of new wiring and gold-plated DL connectors have been incorporated into the technical upgrading of the studio. Construction of a Synclavier pre-production suite has also been completed. "Until now, only those who have owned their own Synclavier have been able to take maximum advantage of the system's amazing capabilities," comments studio manager **Lloyd Donnelly**. "it is our intention to introduce the Synclavier to clients by specifically addressing their production needs at an affordable price." "The acquisition of the SSL console and the Synclavier II has been met with great enthusiasm," states president/chief engineer **Paul Wickliffe**. "Our efforts over the last six years have led us to the point where we are now proud to offer state-of-the-art technology to our clients." *36 West 37th St., New York, NY 10018. (212) 594-7484.*

SKYLINE — custom SSL SL4000 console

□ **SIGMA SOUND** (New York City) has upgraded Studio 8 with a 40-input **Solid State Lofic SL 6000** console with **Total Recall** and special outputs for six-track film mixing. A 52-input **Neve 8078** board with **NECAM 96** automation is to be installed in Studio 7. As a part of the upgrade, the studio is also being equipped with new **Studer A800** multitracks. According to studio president **Joe Tarsia**, the new consoles — which replace custom MCI gear — reflect changes within the recording industry itself. "Sigma is a studio that has always been predominately staffed with house engineers," he notes. "In the past, we tailored our equipment to best suit the needs of that kind of situation. But, as the industry has gone more and more toward freelance engineers, we thought it best to put in equipment that accurately represents the tastes of most freelance engineers." The facility also will be outfitted with a 32-channel **New England Digital Synclavier** digital system. **Barbara Tiesi**, who managed the studio since its inception in 1976, has relocated to France, and will handle Sigma's international business from her new Paris location. She will be replaced by **Hank Myer**, the former operations manager of Radio Band of America. *1697 Broadway, New York, NY 10019. (212) 582-5055.*

Southeast:

□ **RON ROSE PRODUCTIONS** (Tampa, Florida) has opened a new recording facility equipped with two eight-track studios that will specialize in radio commercials, TV voice-overs and slide-film narration. The facility will also house a large collection of music and sound effects libraries. The studios house **Soundcraft Model 600** 16-channel consoles, **Lexicon** digital reverb units, and a **BTX Shadow** synchronizer system. The duplication department will offer high-speed reel-to-reel and cassette duplication in both mono and stereo. *3409 West Lemon St., Tampa Fl 33609. (813) 873-7700.*

NEW AGE SIGHT & SOUND — new facility

□ **NEW AGE SIGHT & SOUND** (Atlanta, Georgia) is a new digital audio recording studio designed by **Jerry Milam** of **Milam Audio**. The 6,400-square-foot complex has two control rooms and studios, plus a video recording/production studio. Control Room A, a LEDE-type room, is equipped with the **Sony PCM-1610** digital processor, a 50-input **Sound Workshop** console with **ARMS** automation, and **UREI 813B** monitors. Control Room B is equipped with an analog 24-track, a 28-input **Sound Workshop** console, and **Tannoy SRM-15X** monitors. Other equipment include an **Ampex VPR-2** for one-inch video recording, and a **Lexicon 224XL** digital reverb unit, as well as the new **Klark-Teknik DN780** digital reverb. Both Studio A and B utilize **JBL 4430** and **4411** monitors. *Atlanta, GA.*

□**ALPHA AUDIO** (Richmond, Virginia) has added **Kim Person** as a staff engineer, according to president **Nick Colleran**. Before joining the studio, Person recorded music sessions in her one-inch eight-track studio. She handled the engineering, arranging, and ed singer and songwriter, who also plays pedal steel guitar, keyboards, and banjo. *2049 West Broad St., Richmond, VI 23220-2075. (804) 358-3852.*

ALPHA AUDIO — engineer Kim Person

□ **SYNHARMONIC PRODUCTIONS** (Orlando, Florida) has opened a new 16-track facility featuring a **Tascam 85-16B** multitrack running in sync with a **Music Data** MIDI sequencer. The studio offers a wide assortment of synthesizers, drum amchines and signal processing gear; other equipment includes a **Mason & Hamlin** grand piano, **Fender Rhodes, Tascam** eight-track and mikes by **AKG, Sennheiser,** and **Shure**. According to studio owner and producer **Michael Davis**, "We offer a wide range of services and place a strong emphasis on efficiency and creativity in a warm and comfortable environment." *728 W. Smith St., Orlando, FL 32804. (305) 422-3444.*

STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

South Central

☐ **DALLAS SOUND LAB** (Irving, Texas) has installed a **Solid State Logic SL 6000 E Series** 56-channel automated console with **Total Recall** to Studio A. A variety of digital outboard gear also has been added, including a **Quantec Room Simulator**, **AMS DMX15-80s** stereo digital delay and pitch shifter, and a **Kurzweil 250**. According to studio manager **Johnny Marshall**, "Although we have only been on-line since January, most of the producers and artists in this area are aware that we have the new equipment and many of them have actually worked on the new console. Since the installation of the new equipment, we have seen a more drastic increase in our business than originally anticipated. Before we installed the console, we usually booked the studio about a week-and-a-half or so in advance, and it was very consistent. Now we are booking more like three and four months in advance." *Three Dallas Communication Complex, 6311 N. O'Connor, Suite N-9, Irving, TX 75039-3510. (214) 869-7657.*

☐ **SOUND EMPORIUM** (Nashville) has taken delivery of a **Neve 8128** console, which has been customized to meet the studio's specific recording needs. The track assignment panel has been incorporated into the overbridge for enhanced operation, while the patch bay has been modified as an external unit to be placed beside the engineer. A total of 44 mike inputs and four echo modules with 32 output channels have also been configured into the 48-channel board. In explaining the acquisition of the new console, **Rick Horton** studio manager says: "It was the a combination of the board's features, the NECAM system, and Neve's longstanding reputation. We also wanted something that would allow us to sync audio to video." In addition to the 8128, the facility also ordered Neve's **NECAM 96** moving fader automation system. The studio will be using the NECAM 96, according to Horton, "in conjuction with our **Mitsubishi [X-800]** 32-track digital machine." *3102 Belmont Blvd., Nashville, TN 37212. (615) 383-1982.*

Midwest:

☐ **TRC MID AMERICA RECORDING CENTER** (Indianapolis, Indiana) has added an **E-MU Systems Emulator** to its complement of keyboards. Also available is a **Yamaha DX-7, Roland JX-3P, Memory Moog, Oberheim FVS-1**, and **DX** drum computer. The **Harrison 3232B** console in the new **Russ Berger**-designed Studio A has received a complete overhaul by staff maintenance engineer **Kevin Van Wyk**. All op-amps were replaced with Harris HA-4605s and Signetics NE-5534s, while the VCAs were replaced with **Valley People ECG101s**. Also available for signal processing is a **Lexicon Super Prime Time, Eventide H910 Harmonizer, Eventide Instant Flanger, ADR** gates, EQs, limiters, de-essers and **Auto Panner, UREI** limiters, **EMT 140** stereo plate, **AKG BX-20E, Lexicon 224** digital reverb, plus **Garfield Electronics Dr. Click**. The monitors are **JBL 4435** bi-amped with a **Crown PSA-2, DC-300A** and **JBL 5234**. The back wall of the LEDE-type room is covered with **RPG Di'Antonio Quadratic Diffusers.** *5761 Park Plaza Court, Indianapolis, IN 46220. (317) 845-1980.*

☐ **KEYNOTE STUDIOS** (Youngstown, Ohio) has added a **Yamaha DX-7** synthesizer and an **RX-15** digital drum computer to its keyboard production studio. An **Otari MX-5050B-II** two-track recorder also has been installed in Studio A. **Gary Kekel** has been appointed executive producer in charge of creative development, and will be heading up the studio's syndicated music division. *4322 Mahoning Ave., Youngstown, OH 44515. (216) 793-7295.*

☐ **WEST BANK SOUND** (Minneapolis, Minnesota) has opened a new 1,400-square-foot facility that features **Tascam** and **Studiomaster** mixing consoles and a **Tascam** eight-track, according to owner **Lawrence Fried.** *1413 Washington Ave. South, Minneapolis, MN 55454. (612) 370-0098.*

☐ **A.R.S. RECORDING STUDIO** (Alsip, Illinois) has purchased a new **Otari 24-track with full remote and locate capabilities. The studio's Trident Series 70** console also will be updated to a 28-by-24 format. *11628 So. Pulaski, Alsip, IL 60658. (312) 371-8424.*

☐ **SWEETWATER SOUND** (Fort Wayne, Indiana) has added a **Kurzweil 250** digital synthesizer, which is offered free-of-charge to customers at the studio. *2350 Getz Road, Fort Wayne, IN 46804. (219) 432-8176.*

☐ **CREATIVE COMMUNICATIONS** (Sioux Falls, South Dakota) has opened what it describes as the first automated 24-track facility in the state. Recently remodeled **Renaissance Studio** (Studio A) now boasts a modified LEDE-type control room featuring a **Sound Workshop Series 34** console with **ARMS** automation, and an **MCI JH-24** 24-track with **Autolocator III.** Stereo mastering is via a **Revox PR99 Mark II**. Control room amplifiers include **Carver M1.5** and **M200** powering **JBL 4430s, 4311Bs** and **Auratone 5Cs**. Outboard gear includes: **Ursa Major Stargate 323** digital reverberation, **DeltaLab Effectrons, Eventide H949 Harmonizer, Korg SDD3000 delay, UREI 1176** and **LA4** limiters, **Orban Sibilance Controller, EXR Exciter, Valley People Dyna-Mite, Omnicraft** noise gates, and assorted microphones by **Neumann, AKG, Sennheiser, E-V, Shure**, and **Sony**. The studio has also added the **E-mu Systems Emulator II** digital sampling keyboard, **Roland Jupiter 6** synthesizer, and **Oberheim DMX** digital drums. *3700 South Hawthorne, Sioux Falls, SD 57105. (605) 334-6832.*

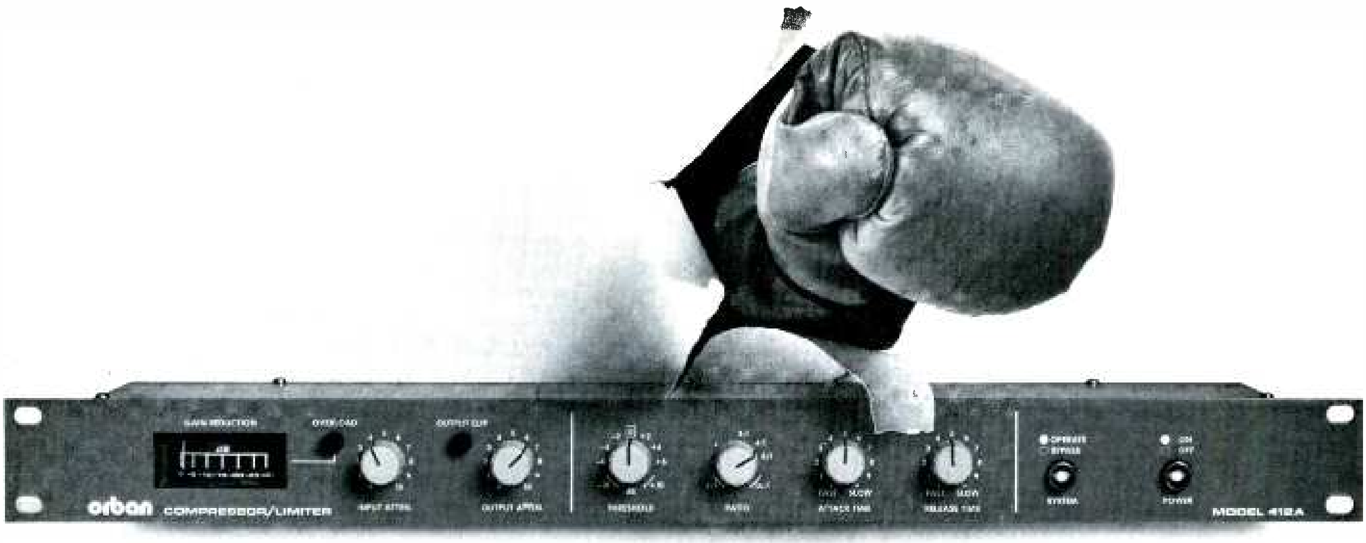
Southern California:

☐ **HAJI SOUND** (Hollywood) is owned by **John Fiore** and **Leigh Straightarrow Fiore**, who have opened a fixed-site location for their 16/24-track recording facility. Fiore, a pioneer in the field of remote recording, began his career in 1969 with CBS Records, and in 1970 was supervisor of recording operations at its San Francisco facility. In 1972 he co-founded Haji Sound, and was joined in 1979 by Straightarrow Fiore as director of production. The studio features a custom **API** console, **George Augspurger** studio monitors, **3M M79** 16/24 track, **AKG BX-20** stereo chamber, **dbx** complement, and **Lang, Pultec, UREI 1176** and **LA3A** limiters. *665 North Berendo, Hollywood, CA 90004. (213) 665-4254.*

LE MOBILE — now based on West Coast

☐ **LE MOBILE** (Woodland Hills) remote recording and mixing truck has moved its based of operations from New York to **Los Angeles**, according to owner and chief engineer **Guy Charbonneau**. The move was made to accommodate a growing demand on the West Coast for the mobile's equipment and capabilities. Le Mobile is said to be the only remote truck in North America with a **Neve/NECAM** automated console and dual **Studer A800** 24-tracks. However, Charbonneau emphasizes that the mobile will continue to be available for projects throughout the United States and Canada. *22240 Victory Blvd., Suite E112, Woodland Hills, CA 91367. (800) 662-4538 or (818) 992-8481.*

☐ **CHEROKEE STUDIOS** (Hollywood) has completed modifications to its four **Trident A-Range** consoles, which has "simplified the function of the British-made boards, while retaining convenience and excellent sound," according to studio principal



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STUDIO UPDATE

FACILITIES EQUIPMENT PEOPLE

Dee Robb. "We got our first Trident A-Range at the AES show in 1976, and began to modify it almost immediately. We wanted to change, among other things, the cue systems, the way the solo systems worked, and the fact that there was no stereo bus. We loved the sound of the board, but we felt we could improve the design." The modifications were designed by **Grey Thompson**, and carried out by the facility's chief engineer, **Toby Foster**. The studio has also launched a new division — **Cherokee Technical Services** — that intends to offer a wide range of technical support services to the local recording community. A full team of technical experts and specialists will be on-call around the clock, seven days a week, and the studio will provide a fully equipped diagnostic laboratory for immediate response to any studio problem. *751 N. Fairfax Avenue, Hollywood, CA 90046. (213) 653-3412.*

☐ **FUTURE DISC** (Hollywood) recently added a **Sony 1610** digital processor and two **BVU-800DB** video recorders to its existing digital gear, which includes a **Mitsubishi X-80** machine. According to president **Gary Rice**, these recent additions provide complete digital and analog mastering services for Compact Disc, record, and cassette manufacturing for the facility. *3475 Cahuenga Boulevard West, Hollywood, CA 90068. (213) 876-8733.*

Northern California:

☐ **C.D. PRESENTS, LTD.** (San Francisco) has opened a 24-track facility in the former **Rhythmic River Productions**' studio. Boasting a 1,350 square-foot recording area with 16-foot ceilings, the control room houses an automated **MCI JH-600 Series** console, **Lexicon 200** and **AKG BX-10** reverbs, along with delay lines and pitch shifters from **Lexicon** and **DeltaLab**. Recent equipment updates include a transformerless **Studer A80MkIV** 24-track, and six additional input modules with parametric EQ. A large isolation booth is currently in the design stage. *1230 Grant Ave., Suite 531, San Francisco, CA 94133. (415) 221-1112.*

☐ **THE MUSIC ANNEX** (Menlo Park) has acquired a **Audio Kinetics Q.Lock 3.10C** timecode synchronizing system. Says **Keith Hatschek**, the facility's marketing manager, "The Q.Lock system allows us to offer a complete, cost-effective package to producers doing audio post-production for video. [Now] clients can leave with a completed soundtrack ready to lay back to their video master, often with a one-day turnaround." *970 O'Brien Drive, Menlo Park, CA 94025. (415) 328-8338.*

☐ **KOVR-TV** (Sacramento) an ABC affiliate, recently purchased equipment from **Sound Genesis**, a San Francisco-based equipment supplier, which will enable the station to effect the transformation from regular mono to Stereo TV broadcasting. The equipment purchased includes an **Auditronics Series 300** audio console; **Orban** stereo generator, an **Optimod-TV** audio processor, and a stereo synthesizer; an **Aphex Compellor**; and audio monitoring equipment, tape machines, and microphones. **Bob Hess**, chief engineer at the station stated: "The technological aspects of the audio field today have become impenetrably complex. What puts Sound Genesis ahead of the field is that they have taken the time to familiarize themselves with this state-of-the-art technology." *1216 Arden Way, Sacramento, CA 95815. (916) 927-1313.*

☐ **PRAIRIE SUN RECORDING** (Cotati) has taken delivery of a **Studer A80** 24-track with autolocator and remote control. Other additions include: a second **AMS RMX-16** reverb, with updated software. Instrument additions include a **Sequential Prophet 10** synthesizer with MIDI control and updates, an **E-Mu Systems Drumulator**, and a **Simmons** drum system. *PO Box 7084, Cotati, CA 94928. (707) 795-7011.*

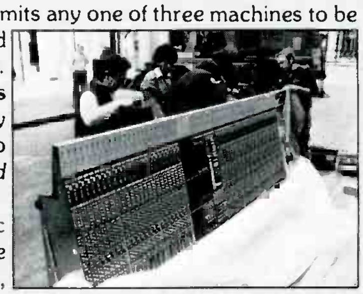
Northwest:

☐ **THE MUSIC SOURCE** (Seattle, Washington) recently finished the construction of an all-new studio designed by **Herb Chaudiere** of **Towne, Richards & Chaudiere**, acoustical consultants. The new control room is approximately 20-by-20 feet and houses an array of new gear including a **Sony/MCI JH-24** multitrack, an **Adams-Smith** synchronizer, and a **Kurzweil 250**. The room is geared for synthesizer production with Kurzweil MIDI-Out going to a **Yamaha DS-7, Oberheim OBXa, E-mu Systems Emulator,** and **Moog** synthesizers. Drum machines include **Oberheim DMX, LinnDrum** and **Simmons**. Monitors are **JBL 4430s** with an 18-inch subwoofer crossed over at 80 Hz. The recording room features three isolation rooms and a "live" section that can be closed off from the main studio with a theatre curtain, which allows substantial tuning of both the "live" and "dead" sections of the room. *615 East Pike St., Seattle, WA 98122. (206) 323-6877.*

Great Britain:

☐ **MAYFAIR STUDIOS** (London) has re-equipped Studio 1 with a **Solid State Logic 6000 E Series Stereo Video System,** fitted with **Studio Computer, Plasma Metering** and 48 channels with **Total Recall**. The new console complements an SL6000 E Series that was installed at Mayfair's Studio 2 in August 1984, and which is also equipped with the Studio Computer and Total Recall, allowing clients to freely transfer their work between both studios. The new SSL is one of the first in London to be equipped with the **Integral Synchronizer** and **Master Transport Selector**, which allows command of up to five synchronized transports directly from the console, with full use of the Studio Computer's locate facilities. It also permits any one of three machines to be selected as the Master, automatically switching all timecode, tach pulse, direction sense, and transport controls and tallies to the computer. *11A Sharpleshall St., London. (01) 586-7746.*

☐ **AIR STUDIOS** (London) has added a 56-input **Solid State Logic SL 4000 E Series Master Studio System** with **Total Recall**. The first SSL was installed in Studio 4 during May 1983. Managing director **Dave Harries** explains that the demands for SSL time in that studio had been "so great that it seemed only sensible to offer another SSL in Studio 2." *214 Oxford St., London. (01) 637-2758.*



AIR STUDIOS — High-flying SSL SL6000

☐ **ROUNDHOUSE RECORDING STUDIOS** (London) has installed a **Solid State Logic SL 6000 E Series Stereo Video System** in Studio 1. The new 48-input console features the **Studio Computer** and **Total Recall**, and is Roundhouse's second SSL System — the first, an SL 4000 E Series with 40 channels of Total Recall, was installed in the Mix Room (also known as Studio 2) in July 1983. *100 Chalk Farm Road, London NW1. (01) 485-0131.*

☐ **MARCUS MUSIC STUDIOS** (London) has added a **Sony PCM-3324** DASH-format digital multitrack, plus a **Solid State Logic SL 4000 E Series** console with **Total Recall** in Studio One. The studio also says it is the first in the world to install **Eastlake Audio**'s new monitors, and has re-designed the acoustics in Control Room One. *49-53 Kensington Gardens Square, London W2 4BA. (01) 229-9595.*

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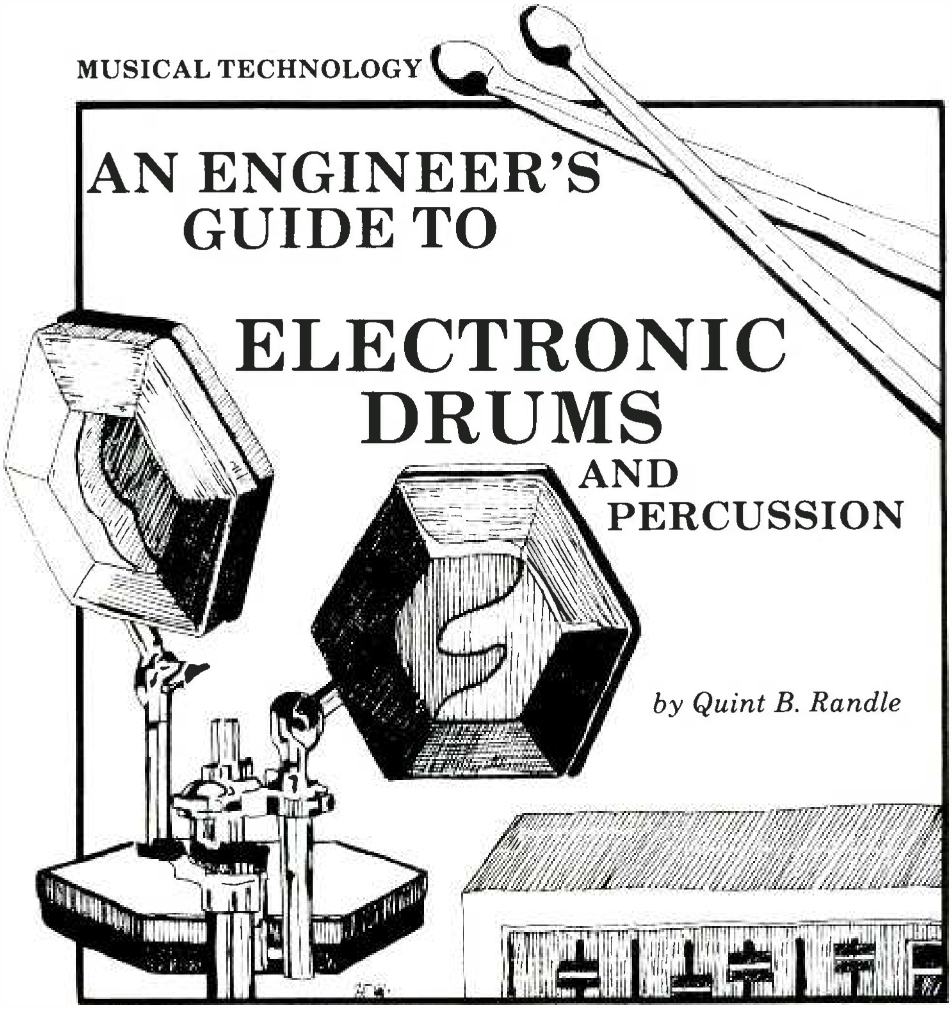
MUSICAL TECHNOLOGY

AN ENGINEER'S GUIDE TO ELECTRONIC DRUMS AND PERCUSSION



by Quint B. Randle

Drum machines, a subject that I dealt with in the February issue, are generally considered to involve the replay of digitally sampled drum sounds that are triggered and sequenced via a built-in microcomputer. The definition of electronic drums, on the other hand, seems to be much broader — for our purposes they can be considered capable of producing analog/synthesized percussion sounds, usually triggered by external playing pads. But a closer look reveals that the line between drum machines and electronic drums is becoming increasingly blurred.

Electronic drums can now be used to trigger digital sounds, or to create combinations of digital and analog sounds. They can also be controlled or triggered by external sequencers and drum machines. It seems that soon —if it hasn't happened already — drum machines and electronic drums will be defined simply as "electronic percussion" since most, if not all, machines and sets will be basically compatible.

There's probably little argument that many studio engineers consider Simmons to be the leader in electronic drums. Simmons does have a hold on the present market, and many engineers and producers have been exposed to the company's products. While other manufacturers also have developed alternate units, some are so new to the market, or have been available for such a short time, that the recording industry has yet to discover them.

As in the previous article on drum machines, the following comments are grouped according to manufacturer and model. It should be noted that if, for one of the above reasons, both myself and the company were unable to locate an educated user, the manufacturer's technical rep was asked for comments.

SIMMONS SDS7

The Simmons SDS7 is probably the most advanced electronic drum kit currently available. Taking up where the SDS5 left off, the new unit has taken the next step and combined both analog sounds and digital samples. Not only are the sound sources mixable, but they can be manipulated to the extreme; each of the unit's 12 channels includes 15 control variables. The SDS7 can also store more than 1,000 drums sounds at on time —99 different sets of 12 drums and cymbals.

Andy Topeka, technical director for The Cars, offers the following advice: "When you start to work with the SDS7, be patient! The same controls that can create an incredible sound can also leave you frustrated and dissatisfied."

Topeka's credits on The Cars' latest album, *Heartbeat City*, included production assistance and Fairlight programming. The SDS7 is not featured on the record, Topeka explains, because it was not available until after the album was finished. Instead, main ingredients of the album's drum sounds are from an SDS5, Fairlight samples and several other things producer John "Mutt" Lange threw in for good measure. The group's drummer, David Robinson, received his SDS7 about a week before the beginning of a tour to support the new album. As a result Topeka's experience is based on use of the SDS7 in a concert setting.

"Triggers attached to each of Robinson's accoustic drums activate the SDS7 to give a synthesized drum sound along with the miked acoustic sound," the engineer explains. Before reaching the SDS7 though, the signal produced by the custom trigger mikes is sent through another custom device that creates a clean, fast trigger for the Simmons. "For a good sound, the SDS7 does require a good, *clean* trigger, so you do need some sort of a box to do a translation."

In other words, the custom box takes a fluctuating signal and reshapes it into a constant, reliable trigger impulse for the SDS7. (The device Topeka and crew fabricated does basically the same job as the Marc MX1.)

Robinson also uses the Simmons playing pad to obtain the sound of the SDS7 by itself, in addition to the synthesized/acoustic live sound. Because the SDS7 replaced the SDS5 immediately before the current tour, Topeka admits that they didn't have to overcome any new problems with the kit, besides "getting a far, far better sound right off the bat." Other than cleaning up the acoustic trigger signal — which is not at all uncommon with electronic percussion — nothing really traumatic happened.

Because the SDS7 is so advanced in the features it offers, by its very nature the device can be complicated to operate. Which is where Topeka's "Be Patient!" advice is founded. One problem that most users, including Topeka, have experienced is losing a sound during the creation process. If you accidentally press an incorrect button while in edit mode, the unit immediately sets you back to the sound you started with — *without* edits. As with any computer-controlled gear, says Topeka, the way to remedy the problem is to save the sound periodically after each individual parameter is changed. "You can't be casual like you could on a Prophet II or DX7,

70% OF?

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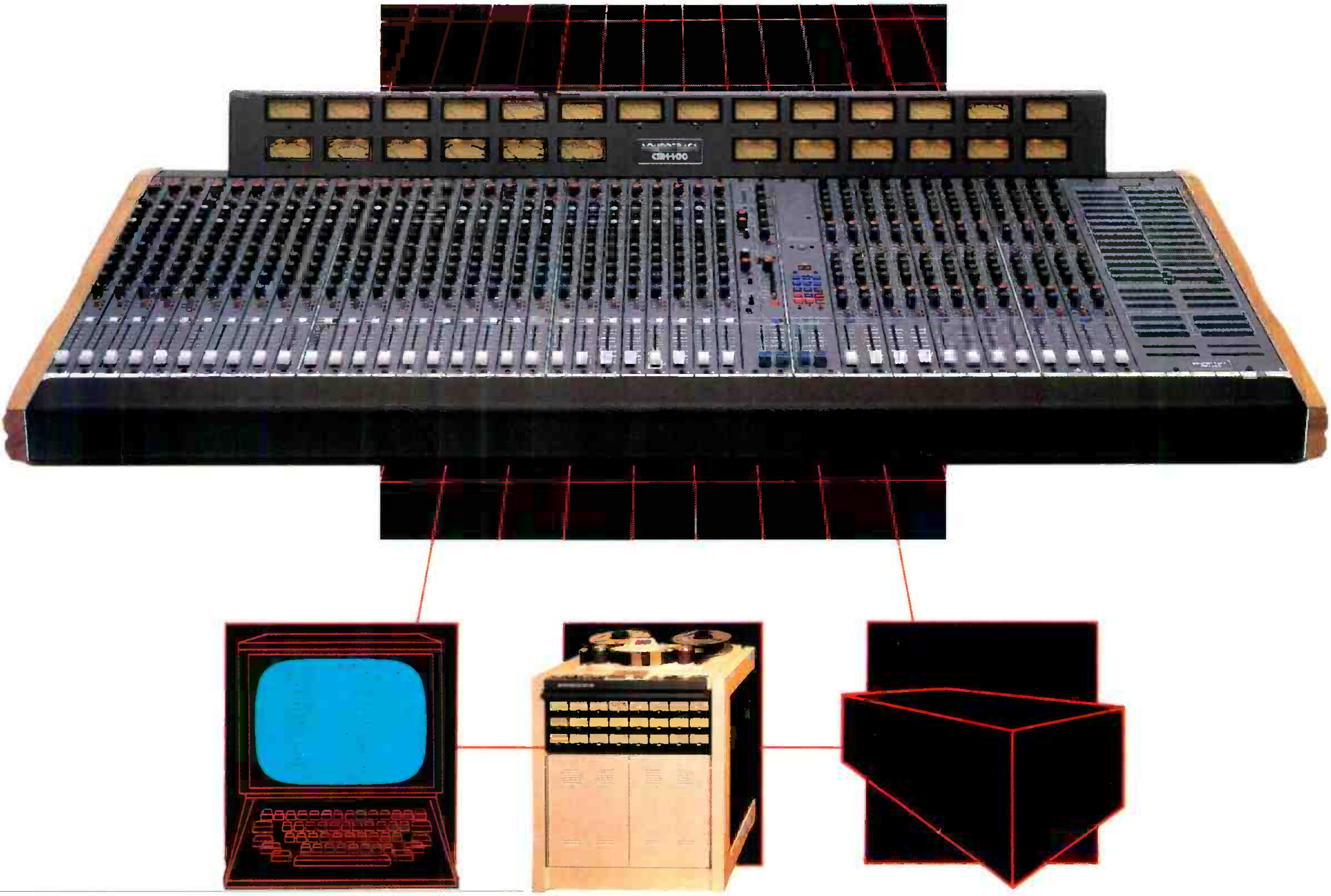

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ELECTRONIC DRUMS AND PERCUSSION

where there's an automatic edit mode or a patch buffer . . . continually save your memory edits as you go."

SIMMONS SDS8

The Simmons SDS8 is something of a scaled-down version of the original SDS5, and is designed for those users that cannot afford the higher price tag of the SDS7. The company has pretty much set *the* standard for electronic drums, so there is little that can be said about the basic Simmons sound that hasn't been said before.

One of the most useful things about the SDS8's owners manual is the way in which each of the controls is explained; it serves as a very useful primer for engineers and producers who are unfamiliar with the variables in electronic drumming. Each control feature is compared in some way to the functions of an accoustic drum. Virtually every electronic drum kit currently available has some of the same features and functions (or close equivalents) found on the SDS8. In understanding these functions — and using an acoustic drum as a common reference point — the programming of electronic drums to sound acoustic or electric becomes a lot more logical.

PITCH: While a conventional drum relies on the tension and size of the drum head to alter its pitch, an electronic drum's pitch can be changed with twist of a knob. (Think of the pitch, tone, or tune control on a module as the tuning bolt on a drum head, only many times more sensitive.)

FILTER: As you strike an acoustic drum, a certain amount of noise is generated. The filter control sets the overall "brightness" of this noise.

— Simmons SDS7 —

Another obvious term for this control is "Noise."

TONE/FILTER BALANCE: Most electronic drum modules feature a control that allows the programmer to create just the right degree of balance between the actual tone and noise (filter). The Simmons manual explains that a timpani would require a lot of noise, and tabla very little.

BEND: The pitch of an acoustic drum starts high and bends downward after being struck. A module's bend control generally determines the amount of bend that will occur — and in which direction — before returning to the original pitch. Tuning the bend upwards creates an effect very different from conventional acoustic sounds.

CLICK: This function is sometimes labelled attack on other kits. Plastic heads fitted to conventional drums create a sharp "click" when struck; this control serves as the electronic equivalent of that part of the drum sound.

DECAY: The time it takes for an acoustic drum sound to die depends on the size of the drum, head tension, etc. The decay control simply tells the synthesized drum how long the sound will last.

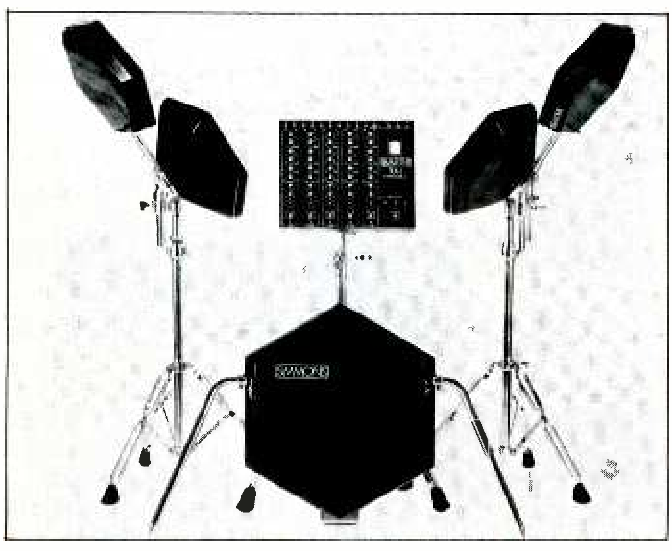
While the SDS8 cannot provide the super-impressive sounds found in the SDS7, it does contain the variables necessary to create the "classic" Simmons sound at a reasonable price — which is what most of the competition is shooting for. Other electronic drum kits may feature most of these controls, but the lack of just one variable may make all the difference between achieving a sound close to *the* Simmons standard, or a cliche "disco sound."

DYNACORD PERCUTER

Dynacord recently introduced an eight-channel digital drum kit that has many of the sound characteristics of a drum machine, but is played mainly with pads. In much the same way that you can change memory chips and sound cards in the Linn and Oberheim machines, the Percuter ("*Per*cussion *Computer*") digital drum computer allows the player to exchange sounds via small, plug-in modules. (These modules can even be changed when the unit is turned on —an important advantage in live use.) Also like most drum machines, the sound in the module itself cannot be altered to a great degree; each channel on the Percuter features sensitivity, volume level and pan controls.

Engineer/producer **David J. Holman** has been using the Dynacord set for several weeks and is beginning to come to grips with the kit's features. He recently completed mixing of the Pointer Sisters' new album, and has five Olivia Newton-John albums to his engineering credits. He was also

— Simmons SDS8 —



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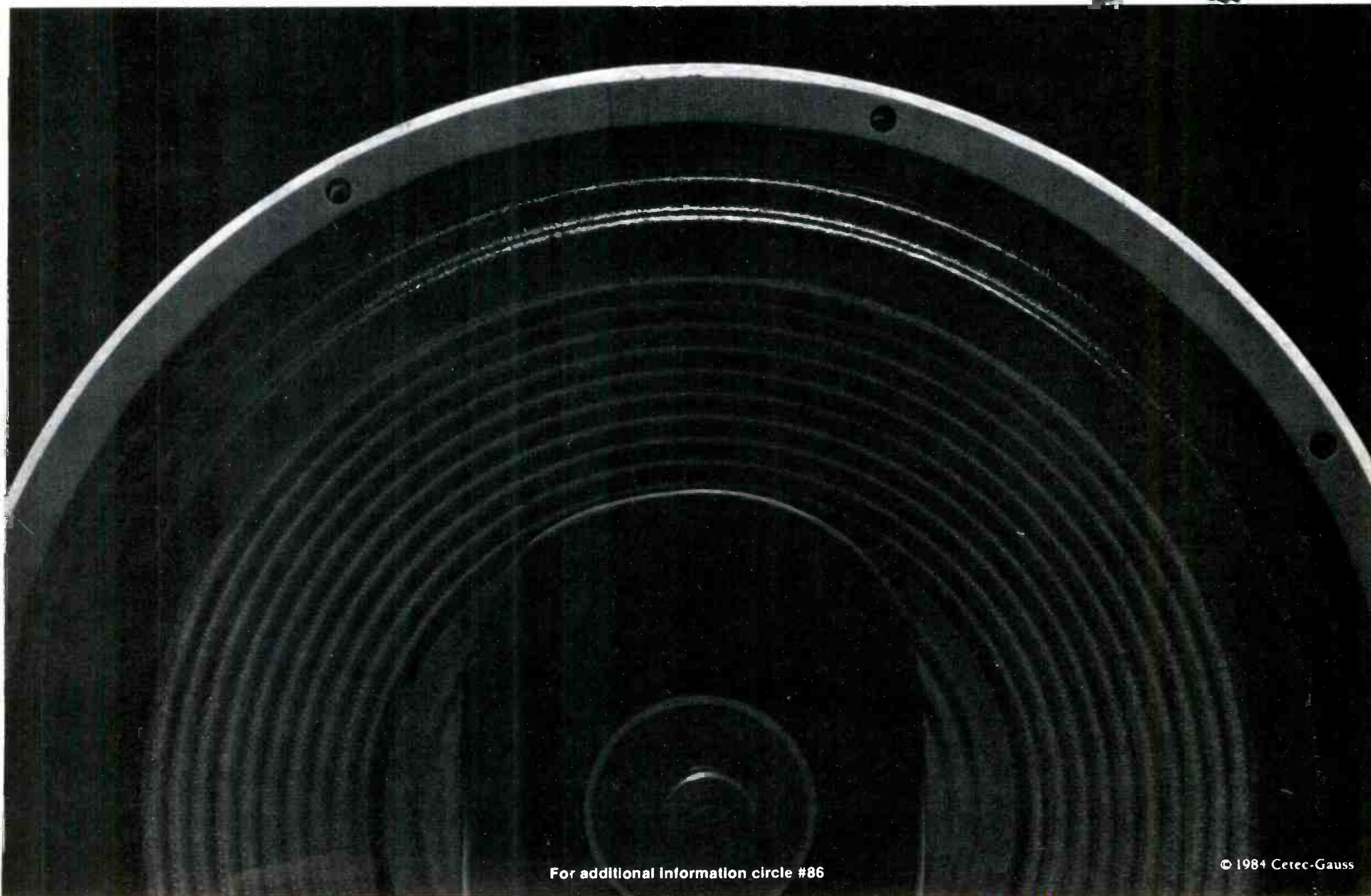

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ELECTRONIC DRUMS AND PERCUSSION

involved with making the recordings used in Oberheim drum machines.

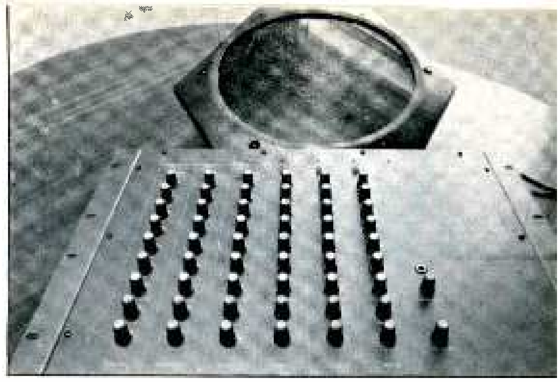
First off, Holman says, the digital sounds are "incredibly real for something you play through pads. It's very punchy and has great bottom end." On a particular session the set provided the perfect feel he was striving for — the sound of a drum machine played live. The sound module library is large, Holman adds, with more than 100 to choose from.

The unit's main drawback, Holman concedes, is that it has "only one tuning circuit." During the same session, the engineer could have recorded the snare and toms in one pass; instead he had to tune the toms the way he wanted, and then record the snare. "It really got in our way with the toms," he recalls.

Larry Oppenheimer, formerly technical manager of Europa Technology, the company that markets Dynacord products in the U.S., says the manufacturer is now including an individual channel-tuning option for an additional $200. So, for little money, Holman's only negative remark about the Percuter can be solved. One technique that Holman uses frequently on any drum sound, whether it be the Dynacord or an acoustic set, can add high- or low-end frequencies without ruining the integrity of the original sound. "Instead of equalizing the drum itself," he explains, "a lot of times I'll side-channel [put onto a second track] the particular drum, compress and EQ it highly, and then add that to the original signal."

This second track will have to be gated, he continues, because by then the noise level of a nearby hi-hat, whether it be high- or low-end, has been brought up significantly. "By compressing the sound, you've made it real tight, but by adding it to the original signal, you haven't taken any punch out of the sound."

Dynacord also makes a sequencer, called "The Big Brain," a unit that enables the user to sample their own sounds into the modules, plus, acoustic trigger microphones designed for use with the Percuter.



— Tama Techstar —

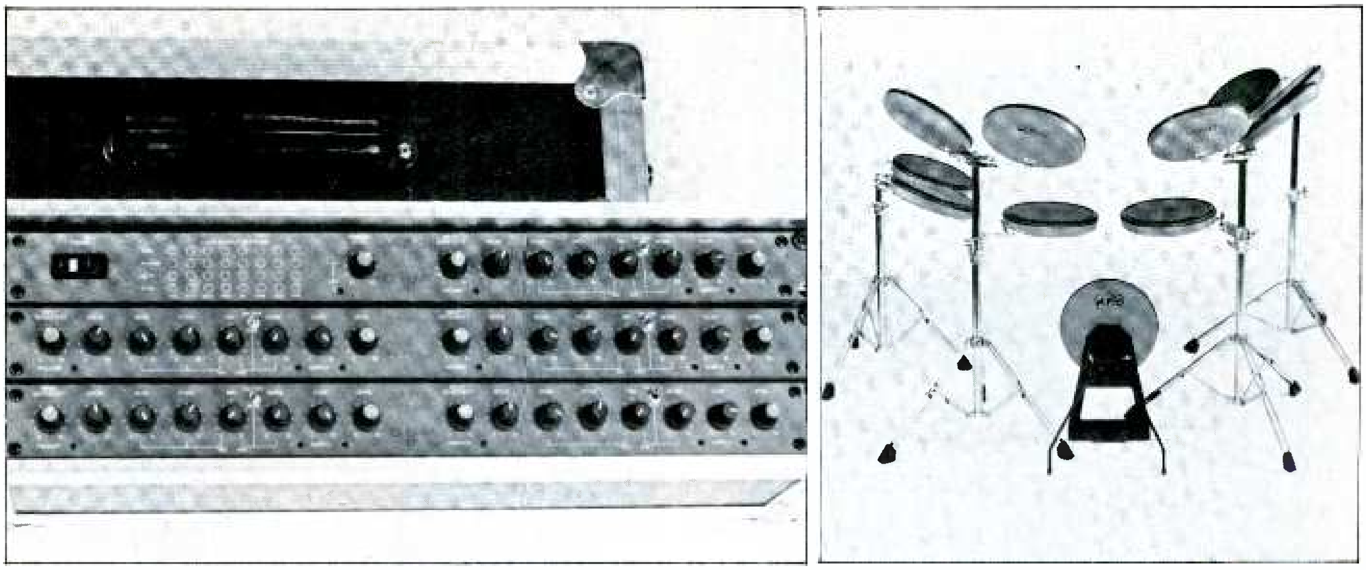
TAMA TECHSTAR

Similar to the Simmons SDS8 is the Tama Techstar Series, which includes the TS305 and TS306, along with the Tama playing pads. The TS305 is the basic unit, featuring bass, three toms, rim shot and snare modules; the TS306 is somewhat of an expansion unit, with four toms, synth and hand clap modules. Controls on the Tama are

TECHNICAL REFERENCE TABLE FOR SELECTED ELECTRONIC DRUMS AND PERCUSSION

Manufacturer/ Model	Simmons SDS7	Dynacord Percuter	Simmons SDS8	Tama Techstar	MPC D.S.M.	C-Ducer "Cactus"	Gretsch ED-700	Music People Drumfire
Sound Source	Digital/Analog	Digital	Analog	Analog	Analog	Digital/Analog	Analog	Analog
Number of Inputs	12 maximum	Eight	Five	Six	One (modular system)	Five (10 maximum)	Five	Five
Pad Input Level	30mV to 500mV	10mV to 10V	5mV to 1.5mV	15mV to 30mV	2V to 5V	300mV to 10V	*	N/A
External Trigger In	Sequencer Input	Sequencer Input	Sequencer Input	500mV to 30mV	5V	Sequencer Input	N/A	*
Sequencer Available	Yes	Yes	Yes	No	Yes	"Available soon"	No	No
Module Control Variables	Modulation Click Content Decay Noise Bend Pitch Modulation Speed Level Filter Functions (3) Sensitivity	Sensitivity Pan Volume	Sensitivity Click Decay Filter Resonance Bend Pitch Tone/noise balance Pan Volume	Sensitivity Attack Decay Noise Bend Tone Tone/noise balance Emphasis Level	Sensitivity Click Decay Noise Bend Pitch Tone/noise mix Level	Sensitivity Decay Noise mix Pitch Filter frequency Filter resonance Pan Volume (Analog mods. have 13 controls)	Decay Pitch Bend Noise Volume	Sensitivity Oscillator decay Noise decay Sweep Pitch O/N balance Pan Volume O/N click
Sounds Saveable	Yes	N/A	No	No	No	N/A	No	No
Stereo L/R Outputs	Yes Preset	Yes User Panning	Yes User-Panning	Yes Panning Preset	No	Yes User-Panning	No	Yes User-Panning
Factory Sounds Included	Digital Samples	Digital Samples	Yes (1)	Yes (1)	No	Digital Sample	No	No
Master Controls	All Individual Controls	Pitch Headphone Volume	L/R volumes Mix volume	Line-in level Headphone level	Volume	Volume	Master and Headphone volumes	L/R Volume
Trigger Devices Available	Playing Pads Sequencer	Playing pads Acoustic Trigger Mikes; Sequencer	Playing Pads Sequencer	Playing pads	Playing pads Acst. Trigger Mikes Drum Computer	Playing pads	Playing pads	Acoustic Trigger mikes
Other Units Available	SDS1 — self-contained pad	Digital Hit	SDS1	Dual module unit	Entire modular system	N/A	Dual module unit	N/A
List Price of Controller	$1,535+ (without sound modules)	$895	$1,077	$750	$315	$1,550	*	*
Basic Set List Price	$4,365 (five channels)	$2,650 (eight channels)	$1,550 (five channels)	$1,299 six-channels	$1,699 (five channels)	$1,950 (five channels)	$1,099 (five channels)	$580 (five channels)

**Information unavailable at press time.*



— MPC Drum Synthesizer Controller (*left*) and Playing Pads —

basically the same as the SDS8, apart from an additional emphasis control. This latter control is similar in function to the loudness button on a domestic stereo system, and can add more "fullness" to a sound.

The Tama pads are a bit less "electronic" looking in appearance, and offer adjustable drum-head tension. While this has no real effect on the sound, it does aid in creating a desired feel for the player. Also, a special trigger block is needed to provide the rim shot sound on the TS305. Both the snare and rim shot channels can be adjusted to sound like toms.

Engineer **Guy Randle** at Rosewood Recording (Provo, Utah) has been using the TS305 for several months now, in addition to the studio's acoustic Tama set, and feels that the unit can sound a lot more acoustic than the Simmons. "When you're out in the hall and you hear a mix going on, the Tamas sound the way you'd like your acoustic kit to sound," the engineer says of the natural sound which he achieves with the TR305.

On the electronic side, he says, the Tama does a good job of producing a full, synthesized sound so often associated with Simmons units. "The noise just doesn't have enough depth to it," comments one session player. "It is kind of a thin white noise."

Although the kick can sound more acoustic than other electronic drums, one drawback is the snare sound, Randle concedes. "It just can't get the electronic 'crack' the Simmons is so famous for." However, the snare is "great for overdubbing to repair a weak acoustic snare," he offers.

The TS305 features both pad and external trigger inputs. While Tama does not sell acoustic trigger mikes, the unit works well with them.

The Techstar Series has individual inputs and outputs for each channel, plus stereo and mono mixed outputs. No panning control is provided on the individual channels; instead, each one is factory set as follows — tom #1 (73%/27% L-to-R), tom #2 (40%/60%), tom #3 (18%/82%), and bass, snare and rim shot (50%/50%).

MPC DRUM SYNTHESIZER MODULES

One feature that sets the MPC kit apart from other electronic sets is that it is completely modular; rack-mountable modules can be purchased one at a time until an entire set is built. According to Bernard Purdie Distribution, the company that markets the English product in the U.S., each module is exactly the same, there being no special channels for bass or snare drums.

The set has been available for about

. . . *continued overleaf* —

— C-Tape Cactus Controller Unit (*left*) and Playing Pads —

a year now, and can incorporate other DSM products such as acoustic trigger mikes, playing pads, and a sequencer. An additional module, called the DSM8+, can provide up to 16 pre-programmed pitch changes as the pad is struck — either increasing or decreasing in pitch, or combinations of both.

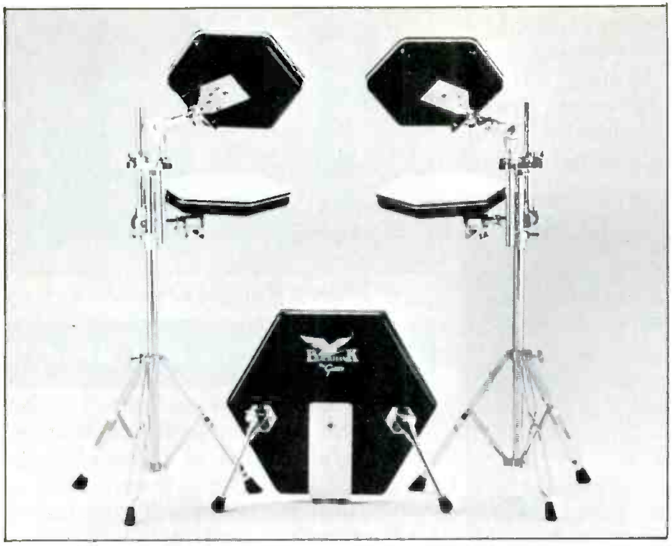
Gary Jackere, percussion manager at Sam Ash Music, a New York music store that handles Simmons, Tama and MPC sets, feels the latter products are underrated: "You're not limited to a fixed setup. Plus, they're as 'ballsy,' if not ballsier than other electronic drums; they have a *very* fat sound." The MPC can sound just as acoustic, if not more so, than Tama's analog set, Jackere says. In addition, the pot adjustment range is much greater than the control ranges on other sets.

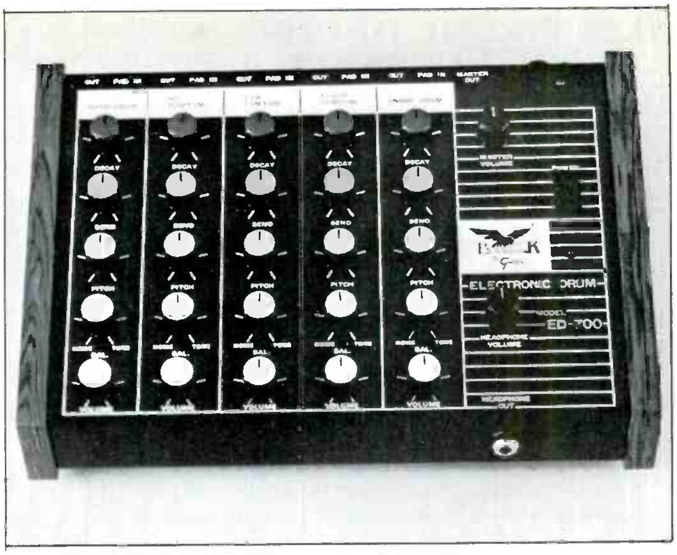
C-TAPE CACTUS

In its basic stock form, the Cactus electronic drum set comes with five pads and a five-channel control unit that contains digitally encoded bass, snare and three tom sounds. The controller can be expanded to 10 channels to provide more digital sounds, including cymbals, high-hat, claps and gong, or analog channels similar to the Simmons SDS8 and Tama Techstar. While the Cactus does not have as many analog-type controls over its built-in digital samples as the Simmons SDS7, it does have most of the key manipulators (see accompanying table). And this is the only kit that offers both digital and analog sound sources in separate modules.

Another feature that sets the Cactus apart is its optional hi-hat module and pedal, which acts in much the same way a conventional hi-hat pedal: depressing the pedal provides a tight sound, while letting your foot off gives a looser sound. Closed-and open-decay pots add more control.

Glenn Mullis, VP at C-Tape Developments, feels the bottom line on the Cactus is that "all other things being equal, it's inexpensive. How can you get into five channels of manipulable digital for $1,950 list?" What you've got to give up in order to get this, he offers, is "fashionable six-sided playing pads" and more control over the digital samples.





— Gretsch Blackhawk ED-700 Playing Pads (*left*) and Controller Unit —

Optional analog modules include: sensitivity; click; tone level, pitch, decay; noise level, frequency, sweep, resonance; pan; volume and feed-through — another click-type feature that adds the acoustic sound of a trigger signal to the sound of the other three generators. Offering a combination of digital and analog sound sources, the Cactus seems to be targeted at users that want digital sounds, but who are torn between the basic, non-manipulable samples of the Dynacord and the price tag of the SDS7.

GRETSCH BLACKHAWK ED-700

Gretsch would be the first to admit that its Blackhawk ED-700 kit is not-going to put Simmons out of business. "We're not going after the Simmons-end market with this product," concedes marketing manager **Karl Dustman.** "It's a learning process for us too."

Terry Wion, who works at Ace Music, a Gretsch dealer in Dayton, Ohio, that also handles Simmons kits, feels that the ED-700 is definitely aimed at the "Mom and Pop" market: "Here's junior wanting to go electronic and the only thing he can say is 'Simmons.' And the Gretsch set is sitting there for $995, out-the-door."

Mark Richards of Gretsch says that although the ED-700 can produce disco-type and traditional Sim-

ELECTRONIC DRUMS AND PERCUSSION

mons sounds, the quality is not as defined. "You're limited to a preset attack level. To add an attack control would have cost too much, and taken us out of the market were trying to get into," As such, the Gretsch ED-700 represents a good starter set drummers, and would potentially be of interest to small-studio owners.

ANATOMY OF AN ELECTRONIC DRUM SET

While the approach is different, the purpose of an electronic drum kit is basically the same as its acoustic counterpart: to create a rhythmic backbone that drums have been providing for thousands of years. Finally, technology has done for drums and percussion what amplifiers and synthesizers have done for the guitar and piano.

An electronic drum kit can be thought of as being made up of two basic parts: the Trigger, and the Control Unit — both of which can take varying forms.

The Trigger

The most common electronic drum trigger is the playing pad. While cosmetics may differ from manufacturer to manufacturer, the pad's primary purpose is to house the transducer — generally a piezo-electric pickup protected by polymer, foam, or thin sheet metal — or a traditional drum head. Located in the center of the pad, the pickup simply translates the mechanical energy of the pad being struck into an electronic impulse, whose level ranges from four millivolts to 10 volts (see accompanying Table for actual values). Since the voltage generated is dependent upon how hard the pad is struck, a drummer's playing technique controls somewhat the volume and characteristics of the sound.

Other ways of triggering a control unit are limited only by the user's imagination, provided that the signal is within the range of the trigger or pad input level. Other common triggers can be categorized as follows:

Acoustic Trigger-Microphone: These are "portable" piezo-electric pickups that can be attached to acoustic drums (or anything for that matter) using Velcro or some other adhesive material, allowing electronic drums to be used in conjunction with, or to replace, miked acoustic drums.

Drum Machine or *Sequencer:* Since the sensitivity range on most control-unit channels is very wide, triggering in this manner is usually uncomplicated, and no gating will be necessary. You simply take an individual output from the drum machine or sequencer, and feed it to the trigger input on the control unit.

Gated Microphone: When using a standard microphone to trigger a control unit, it is important to gate the signal, thus avoiding false triggering from other drums besides the one being miked. (The gate should be set to open when the mike signal reaches the range of the trigger input on the control unit.) Gating may not be necessary, however, when one drum is being miked and recorded by itself.

Signal From Tape: When triggering from tape, the techniques described above for live miking should be used. Where a particular acoustic drum is not isolated, an additional device can be used to prevent false triggering, and to provide a clean, quick signal for the control unit. These devices are said to allow a more constant and pleasing interface between electronic control units and non-pad trigger devices.

Whatever particular form it takes, the trigger does the same thing to the electronic drum control unit as the keys on a keyboard do for any synthesizer.

The Control Unit

The drum synthesizer or control unit contains one, or a combination of, the following two sound sources:

Digital Sound — When a voltage is received from the triggering device, a digital recording of a particular drum sound is played back. The recordings are held in programmable read-only memory chips (PROM), in much the same way as the sounds in a digital drum machines are stored. The sound is limited only by the storage capacity of the individual PROM.

Analog Sound — The other type of sound source, which is used separately or can can be combined with the digital sound source on some models (namely the Simmons SDS7,) produce drum sounds in a manner similar to the way analog synthesizers create "piano" sounds. A voltage-controlled oscillator generates a tone, the characteristics of which can be manipulated by filtering, changing the decay pitch, etc. (See accompanying article for a fuller explanation of basic analog controls.)

In addition to this, white noise can be added through the use of a tone/noise balance control. These two sources can also be amended by a third source: the click or attack generator, which adds somewhat of a "crack" to the sound.

The control unit, which may contain one channel of analog or 12 channels of digital-analog sound sources, usually includes individual channel outputs, and an on-board mixer that provides stereo- and mono-mixed outputs. The pans are either user-adjustable or pre-set by the factory.

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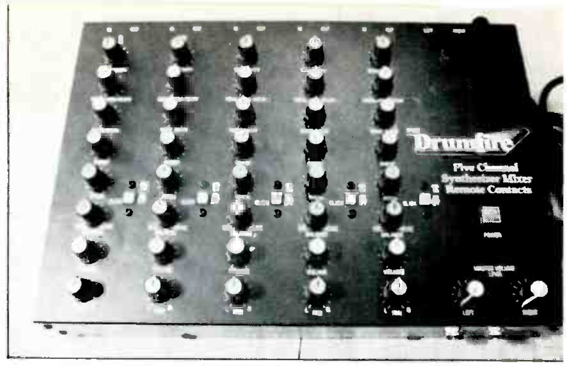


— Music People Drumfire —

MUSIC PEOPLE DRUMFIRE

While Gretsch is going after the lower end of the electronic market with a full set, The Music People (Hartford, Conneticut) is marketing a control unit and acoustic trigger-mikes for those working on a tighter budget. The Drumfire mixer/synthesizer system can be activated by trigger-mikes attached to acoustic drums, or even practice pads — in addition to a signal from tape, sequencer or drum machine.

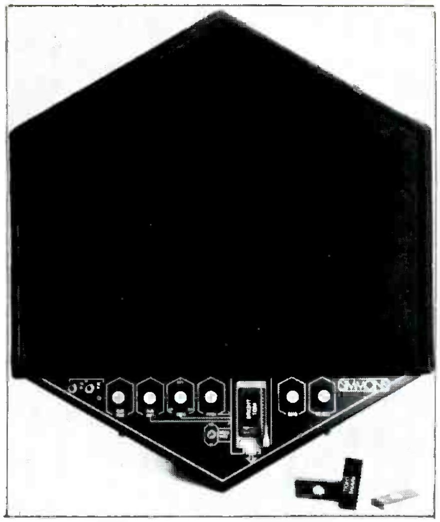
Drumfire offers a legitimate way to create stand-alone synthesized sounds or, more likely, to enhance an acoustic sound. One player I talked with, who triggers the set from practice pads via the attached trigger-mikes, says that the tom sounds are very close to "the Simmons sound," and the snare sound is very pleasing.

When used to enhance an acoustic set, the piezo-electric pickups are generally fastened (using Velcro) to a side of the drum just below the head. Such an attachment provides the greatest amount of mechanical energy to activate the trigger, but does not adversely affect the drum's sound. (If the electronic sound is to be used by itself, the pickup can be placed on the head itself. While this form of attachment deadens the sound of the acoustic drum, it does provide a more reliable trigger source.)

After examining the set for several hours in a studio setting, the main problem I encountered was an occasional slight crackling sound from the trigger mikes. A company spokesperson said the problem is being eliminated with the inclusion of a higher quality trigger mike. If your acoustic set doesn't have enough punch, nor does your pocketbook, then Drumfire may be the way to go.

ODDS and ENDS

Most of the aforementioned manufacturers and others are marketing one-sound/one-pad and dual units for reasons of affordability and practicality. After all, why buy an entire set when you can get by with just a smaller control unit and two pads? In this context, several of the following products might be worth considering:



— Simmons SDS1 —

The **Simmons SDS1** is a self-contained digital drum pad which, like the SDS7, allows the player to change sound chips. Controls include pitch, bend, run time, sensitivity, volume and a trigger input. It can be powered by four 1.5-volt batteries, or an optional AC adaptor.

E-Mu Systems has been selling the **E-Drum** as a single-pad unit, but some players are buying four or five of them to create their own set. The drum is self-contained, and powered by a nine-volt battery or optional AC adaptor. Digitally-sampled sounds can be interchanged using different mini-cartridges. At present, the sound library consists of 14 sounds, ranging from basic percussion sounds to grand piano. Pitch can be changed in three ways: a standard pitch pot; and option that changes pitch according to how hard the pad is struck; and a voltage-control pedal. Other controls include bass EQ, treble EQ, decay, volume and trigger in.

Tama has introduced a line of dual-module **Techstar** systems, called the TS200 Series, which feature the same controls as those found on the TS305 and TS306, but can be purchased in pairs: snare/rim shot; synth/hand clap; or two tom toms.

Gretsch is also taking a step upwards into the dual-module market. The sound of the attack is said to have been improved and increased, and an input sensitivity control added.

Dynacord Digital Hit is a compact device designed to reproduce one digital drum sound. The unit can be triggered by a pad, trigger mike, sequencer, or standard microphone. When the latter input is used, both the digital sound and the live sound can be combined internally to provide a mixed output. While at present the unit's sound chips are not user-changeable, Europa Technology offers that this may change in the near future.

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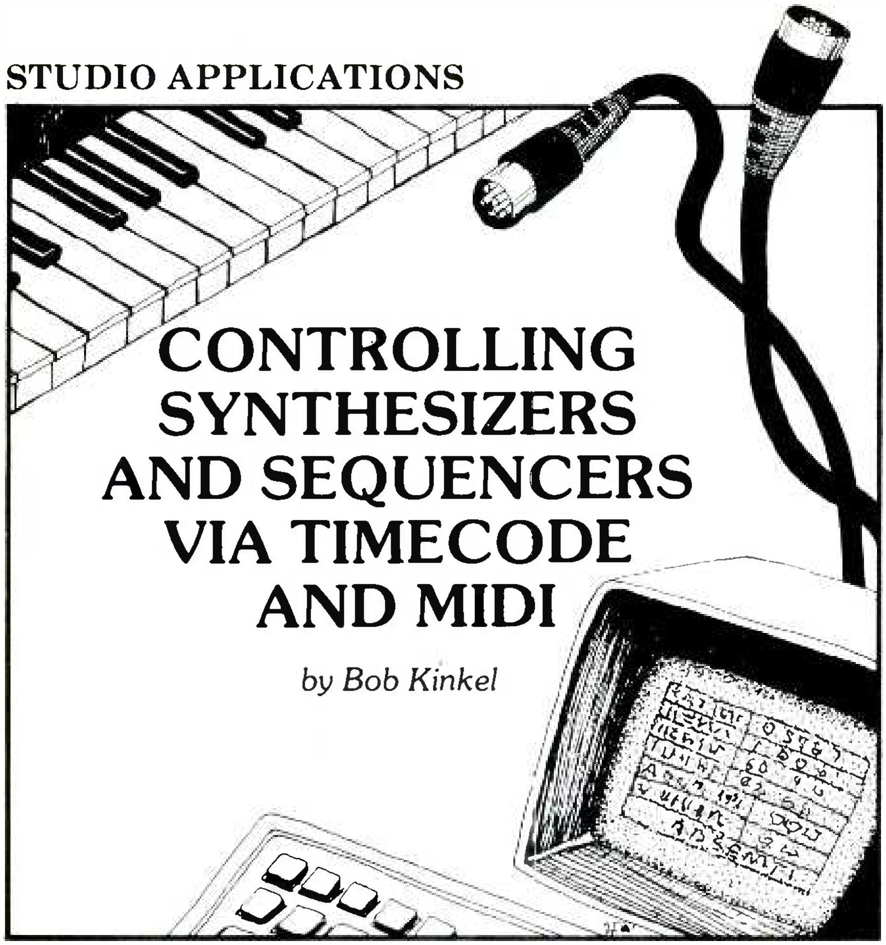

CONTROLLING SYNTHESIZERS AND SEQUENCERS VIA TIMECODE AND MIDI

by Bob Kinkel

As anyone working in the recording studio has probably noticed, these days SMPTE timecode is being used more and more — and not just for autolocators, console automation, synchronizing multitracks, and video editing. With the recent development of timecodereading devices, synthesizers, drum machines and sequencers can now be synchronized with one another to a high degree of accuracy. Engineers no longer have to sweat over how many tape tracks they will loose to conventional sync tracks; one timecode track can be used to run everything.

Timecode-reading sync boxes — including the Roland SBX-80 and Friend Chip SMPTE Reading Clock (SRC) discussed here — translate timecode locations into measures, beats, sequencer/drum machine clocks, and MIDI Clocks, all of which allow the user to know *exactly* where they are in a piece of music. In this way, drum machines and sequencers can be started from *any* point on tape, instead of always counting off from the beginning, as with conventional sync techniques.

Friend Chip SMPTE Reading Clock

The SMPTE Reading Clock, manufactured in West Germany by Friend Chip, and distributed in the U.S. by Europa Technology, Inc. generates and reads 25- or 30-frame SMPTE timecode. It has a programmable tempo in beats per minute that can be used manually (in Free mode) or slaved to timecode. Start time is programmable in HRS:MIN:SEC;FR (when slaved to timecode), and is the point used by the unit for all measure and beat calculations.

A programmable Cue Point is provided for starting synchronization from any beat in the middle of a tune; the SRC will drop into sync on the next metronome beat following any timecode number you enter. The number can be entered manually via the keypads, or the cue point caught on the fly by hitting the cue button while the tape is rolling. The unit also has the nice feature of being able to manually start sequencers anywhere —the clock outputs will start on the beat following your hit of the button, and stop on the beat after you hit it again.

The SRC has two Clock Modules, each with adjustable clock rates, pulse widths, voltage, "+" or "-" leading edge, and different enable (ENA) outputs. The left-hand Clock Module offers slower clock rates (four to 384 pulses per whole note), allowing it to be used as a click-track or an arpeggio-generator (four pulses — ¼-note or click; six pulses ¼-note triplet; eight pulses ⅛-note, etc.). Clock Module #2 has higher clock rates (16 to 768 pulses per whole note clock) for driving Linn, Oberhiem, Roland, Synclavier, and other units. The SRC also has a TTL module that is capable of even higher clock rates, allowing synchronization with Fairlight (1,536 pulses per whole note), and features simultaneous clock and enable (ENA) outputs.

[To avoid confusion, the SRC has clock rates labeled in pulses per whole note, while most manufacturers notate their clock rates in pulses per quarter-note. For example: a 24 pulse per quarter-note sync is written as 96 pulses per whole note; a 48 per quarter as 192 per whole, etc.]

There are two Delay Modules on the SRC that can be patched into Clock and TTL Modules. Each offers a sweepable delay of 1 to 64 milliseconds, a value that can be changed by ear while tape is rolling (Jitter function) to correct processor timing errors inherent to each piece of equipment. (Longer delays can be programmed by changing the start time.)

The SRC's operational advantages can be summarized as follows: multiple clock outputs, with the capability of delaying each in real time; and the ability to adjust clock-pulse width, voltages and phase, which allows the unit to synchronize virtually anything. The only disappointment is that there is no MIDI Clock output.

Roland SBX-80 Sync Box

The SBX-80 generates and reads 30- or 25-frame timecode, and has three sync outputs: MIDI Sync, Roland (DIN) sync, and an adjustable clock from 1 to 120 pulses per quarter-note. Tempos can be entered and read in either beats per minute (B/M), or video-frame bits per beat (F/B), and can be programmed for ranges between 20 and 250 B/M or 90.0 and 7.16 F/B.

Two modes of operation are available: manual and programmed. In manual mode, the SB-80 is a master tempo clock, allowing the tempo to be varied in real time by turning the tempo knob, or tapping it in with the tap button. The unit can also be slaved to an external audio source (click-track, cowbell, etc.). In programmed mode the SBX-80 can record and play back a steady or varying click-track, and play it back slaved to timecode or its own internal clock.

Time signature is programmable in 1 to 15 beats per measure, and the timing value of each beat can be either a quarter- or an eighth-note. While the number of beats per measure can vary

-the author-
Bob Kinkel is a New York-based synthesist, songwriter, composer. He has a strong, technical background, and has dealt extensively with timecode both on the road and in the studio. He is the co-author of "Recording Original Music and Sound Effects for a Videodisk Project," published in the October 1984 issue of *R-e/p*.

— Friend Chip SRC SMPTE Reading Clock —

from measure to measure, the timing value cannot. There is also a count-in feature, which works in either manual or programmed mode, and gives two measure of clicks before the sync starts — a very useful feature if you're printing a click-track for live musicians to play to.

In Record Mode the SBX-80 can read timecode and record tempo data from a click on tape or from other audio input (kick and snare, or cowbell) — which is useful when syncing to pre-existing tapes. It will remember where each click falls relative to the timecode location, and allow sequencers to synchronize to the tape following all subtle tempo variations. In situations where there is no click recorded on tape, tempo data can be programmed by hitting the tap button with the music. If drum-machine or sequencer sync is available, you can print timecode, set up the drum machine or sequencer to produce a click-track, slave it to its sync, and feed the click into the SBX-80 while its in record mode with external timecode switch enable. Tempo Data can be dumped to tape, and the original click-track or sync erased.

The SBX-80's Edit Mode is one of its most useful features: it allows you to correct the timing of recorded tempo data — i.e., you can move a beat that is out of place backwards or forwards a little — or enter the tempo data for each beat from scratch. Being able to program the tempo for each beat is very useful for scoring work (more on this later), and adding musical touches (retards, accelerandos, etc.) to sequenced songs.

— Roland SBX-80 Sync Box —

When playing back the recorded tempo data clocked to external timecode in Play Mode, the SMPTE offset displays the start time of sync in HRS:MIN:SEC;FRS:BIT. This time can be changed to move the sync ahead or behind the beat, and is particularly useful for correcting timing errors. Also provided is a Set Sync function that will allow syncing from the start of any measure in the song. For MIDI-equipped devices capable of receiving such data, song-and measure-select pointers are sent to allow them to sync from any point on tape.

Once you master the owner's manual — it's a bit confusing, and takes a few readings to understand — the SBX-80 is a pleasure to work with. Its advantages are its low price and capability of playing back a varying click when locked to timecode. However some limitations are a programmable Clock Out that does not go high enough to drive a Fairlight CMI, and does not produce simultaneous sync for running many devices; and that there is no way to change the timecode offset while the tape is running in order to fix timing problems by ear — instead, you have to use trial and error, which can be very frustrating.

Applications on a Recording Session

When using these devices on a session, it is important to make all your production decisions *before* the session starts. Once it is laid down, you can't cut the tape and still use the timecode, because each edit will leave a large timing gap that sync boxes and synchronizers will not be able to handle. Stripe the multitrack with continuous timecode from start to finish. Most people print timecode on track #24 at -10 VU; -3 VU works well on semi-pro gear, and multitrack cassette recorders.

Program the offset — or start time for the clock outputs — to begin on an even frame number at least 20 seconds after the start of timecode. The 20 seconds of free-running code leaves plenty of time for the sync boxes to lock up, and is sufficient for slave tape machines to synchronize to the master (should you decide to go 48-track, or are working with video). Find an acceptable tempo, and program it into the unit's tempo memory. On the SBX-80, use the edit mode to enter the tempo data for the song by clearing the memory, program in the first two measures, and use the copy function for the rest of the song. You are now ready to start laying down tracks: wind back the tape to the beginning of the timecode, and roll tape. When the multitrack reaches the programmed start (offset) time, all the sequencers and drum machines will start running.

Note: if you've programmed the SBX-80's auto countoff, the sync will start two measures after the entered timecode offset. To have sync start on the proper code location, it is necessary to set the start time *before* the actual start of sync. To find the new start time proceed as follows: put the SBX-80 in edit mode; select the start of measure one; and press the display button until it displays SMPTE clock. The number shown in the display is

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TIMECODE CONTROLLERS

the amount of time (to the bit) that the count off uses. Subtract this number from the actual time you wish the downbeat, and then enter your new number as the offset time. This is not critical in regular music recording, but its essential for video.

All sequencers and drum machines have inherent timing delays when responding to external sync signals. Because of this delay, you may notice that MIDI-based sequencers lag slightly behind a drum machine being driven by a sync pulse. To eliminate such timing problems, it is necessary to use delays and/or offsets. The SRC features delay modules to correct for timing delays in real time —the synced units can be made to line up by delaying the sync of the slower of the two, so it locks with the other.

The SBX-80 does not have such a delay feature, which means that you have to print the first track and overdub the part that's not in time by moving the timecode offset forward or backward. If what you're syncing is behind the beat move the offset back a few bits; if it's ahead, then move it forward a few. Timecode offsets can be converted easily to real time: one frame of 30 fps timecode equals 33.3 milliseconds (one frame of 25 fps equals 40 milliseconds). Since there are 80 bits to each frame, a one-bit offset equals 0.42 milliseconds (0.5 milliseconds in 25 fps code). I have found that an offset of 30 bits (12 milliseconds) will make a synth bass line fall in the pocket with a drum machine synced from the programmable clock output from the SBX; longer offsets are sometimes needed for synth sounds that have slower attacks.

The usefulness of timecode-driven sync boxes becomes even more apparent when you want to start synchronizing from somewhere other than the beginning of a song. Suppose you want to add a sequencer part that only occurs in the choruses; instead of programming in blank measures, and starting from the top of the tune, you can program the sequences to start at the beginning of each chorus. To do this use the cue point on the SRC, or the Set Sync function on the SBX-80, to select the measure or beat at which you want to start synchronizing. Then, program in the start point, wind the tape to just before that point, roll tape, and everything will start syncing from the downbeat of the measure you select (any beat for the SRC). The SRC has a Manual Mode that allows you to hit the cue button just before the spot where you want your machines to begin playing, and will auto correct your "Punch-in" to start everything in sync on the next metronome beat.

If your sequencer and drum machine can be driven by MIDI Clock, and are able to read the MIDI Measure Pointers, with the SBX-80 you have the added ability of being able to start the tape anywhere you want, and have all your sequenced material drop-in on the next downbeat. This feature is a great time saver if you're using drum machines and synthesizers "live" to save tracks during overdubs or the final mix. You won't have to wind back to the beginning of the tape for each pass — all the machines will drop-in on the downbeat of the next measure after the point at which you roll tape.

Applications in Video Scoring

When working with video it is essential to make sure the timecode you lay down on your audio tape is an *exact* duplicate of the code on the video reel. If, for example, you are adding incidental music to an audio reel that has already been started, you don't have to worry, since the timecode has already been taken care of. In this case, all you have to do is take a timecode output from the multitrack, connect it to the sync box, and start laying down your tracks. When starting a fresh reel, however, make sure you regenerate the timecode from the videotape onto the audio reel, to ensure compatibility and eliminate the need for offsets.

The ability of sync boxes to clock their tempos directly to timecode makes the life of a video composer a lot easier, from pre-production to the final recording. Clocking to timecode ensures that each segment will be played at exactly the programmed tempo, and will always lock to picture (something that is not *always* guaranteed with a free-running clock). Having a programmable start time enables the music to start exactly on the timecode frame it is supposed to, which eliminates the need for any offsets in the post-production mix. Sequencers and similar devices can be synced directly to any tape that has timecode on it. For example, a composer that has a videotape with longitudinal timecode printed on it can check his click and sequenced music directly against the video material, making sure that all events and hits are scored accurately before committing to the score, and laying down the track.

The SBX-80 excels as a video scoring aid. Its ability to program the click for each beat (edit mode), allows for scoring possibilities not feasible with a fixed-rate click. For example: You're working on a scene that opened with a series of establishing cuts, and you think that single chords hitting on each cut work best. With a fixed-rate click you would have to use your timecode edit list to figure out on what part of which beat of which measure each cut would fall, and program it into your sequencer. With a variable click, however, all you have to do is calculate how many frames there are between cuts and use the differences as the tempo for each of the first few beats, allowing the sequencer to be programmed as a series of quarter-notes, instead of the previous confusion.

Another use of the variable click would be for a scene that cuts between two characters or locations, each of which has its own theme music and individual tempo. Musical segments can be made to fit exactly by tailoring tempos a little, or a lot.

Flying in synthesizer effects is also simplified. If you want a "woosh" that lasts from timecode location A to location B, you program one quarter-note in the sequencer, program the start point at timecode A, and the tempo for that beat at timecode (B-A) frames per beat. Roll the tape and it's done.

Every year, people discover new ways of using timecode to simplify life in the recording studio. Until now these timecode synchronizers were only used with tape machines and consoles. The Friend Chip SRC and Roland SBX-80 are the first of a new generation of devices that use timecode to aid directly in the creation of music. Many more are on the way.

■■■

MIDI UPDATE

A Report on the MIDI-Equipped Synthesizers, Sequencers and Software on show at the Spring NAMM Exhibition

by Bobby Nathan, Unique Recording Studios

Of all the new and innovative products utilizing the MIDI interface, none were more striking than the various pitch-to-MIDI converters to be seen at the recent Spring NAMM Exhibition, held at the Anaheim Convention Center in early February. **Fairlight** and **Cherry Lane Technologies** (for IVL Technologies) were the two companies showing such "wonder boxes."

The Fairlight Voicetracker can take any monophonic audio source and turn that signal into MIDI data which, in turn, can play any MIDI synthesizer and/or can be recorded into any MIDI sequencer. For example, a most useful studio application would be to take a lead vocal track from tape, and during mixdown have a synthesizer set to a string patch follow the pitch and envelope of the same vocal track. But that's only the beginning: Plug in any microphone, and voice, guitar, sax and virtually all monophonic instruments can simultaneously control *any* MIDI synthesizer to reproduce whatever riff is being played into the Voicetracker. The unit can display the input waveform on a video monitor (not supplied) in either a vertical or horizontal scrolling format. A zoom function is also supplied for taking a closer look at the waveforms. There are different displays for pitch, brightness, amplitude, purity and pitch error. The Voicetracker also supports analog synthesizers via CV and ga. outputs.

The Pitchrider 2000 from IVL Technologies Ltd. (and distributed in the U.S. by Cherry Lane Technologies) performs the same basic functions as the Voicetracker, in relation to taking any audio source and converting it to MIDI. (The device was originally developed as an educational aid to help musicians improve their intonation skills.) The front panel simultaneously displays alphabetically the pitch of any note that is input into the Pitchrider 2000 (Ab, C#, etc.), and in which of six octaves. Even though the Pitchrider does not offer video display modes — which might only be necessary if you live near the San Andres Fault, and put a microphone to the ground — it does the job of turning monophonic pitch into MIDI control parameters.

As I had mentioned in previous articles on MIDI-equipped devices, we have only begun to see the effects that MIDI will have on outboard gear on-stage as well as in the studio. **Roland** and **Akai** were two manufactures showing MIDI-capable outboard products.

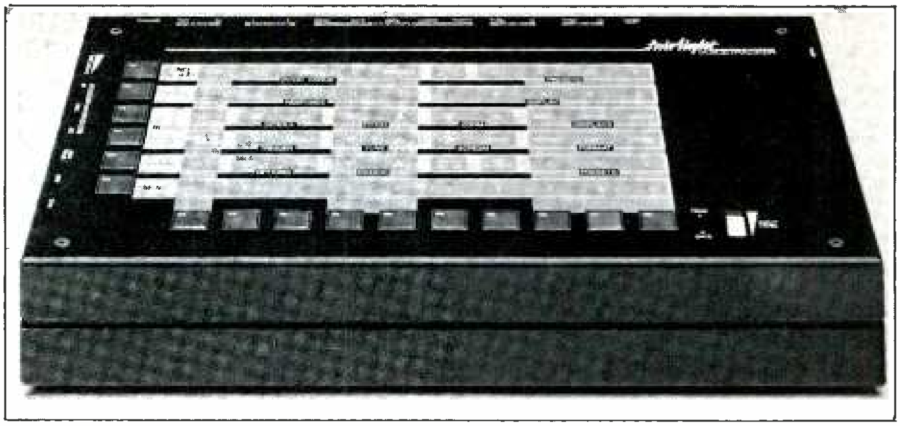
The Roland SRV-2000 digital reverb can store 24 separate combinations of pre-delay time, reverb time, high frequency damping, room size, equalization, gate time and output level controls. Program changes can be coordinated with most

MIDI UPDATE

MIDI synthesizers and sequencers.

Roland also introduced the SDE-2500 digital delay that offers a maximum delay time of 750 milliseconds, and a quoted dynamic range of 90 dB with 0.05% THD. Between zero and 10 milliseconds of delay can be set in increments of 0.1 milliseconds for fine tuning; over 10 milliseconds the delay is set in one-millisecond increments. The SDE-2500 can store 64 different presets of delay time, feedback, modulation depth and rate, plus dry/delay balance. Again, the unit can be interface with other MIDI equipment for automated outboard effects.

Akai made its debut in the outboard gear market with the S612 digital sampler, which can sample for up to eight seconds at a 40 kHz sampling frequency. The unit also features truncation of the sampled sound at the front as well as the rear of the sample, plus modulation and triggering from the audio input, as well as from any MIDI-equipped keyboard. In addition, when played from a MIDI keyboard the sample becomes six-voice, and can be transposed over the full range of MIDI. To allow for greater flexibility, any note can become the original pitch of the sample in transposition mode. The S612 also supports an optional Commodore cassette interface, and/or an optional "quick-disk" disk drive can be added for storage and retrieval of your sampled sounds.



— Fairlight Voicetracker —

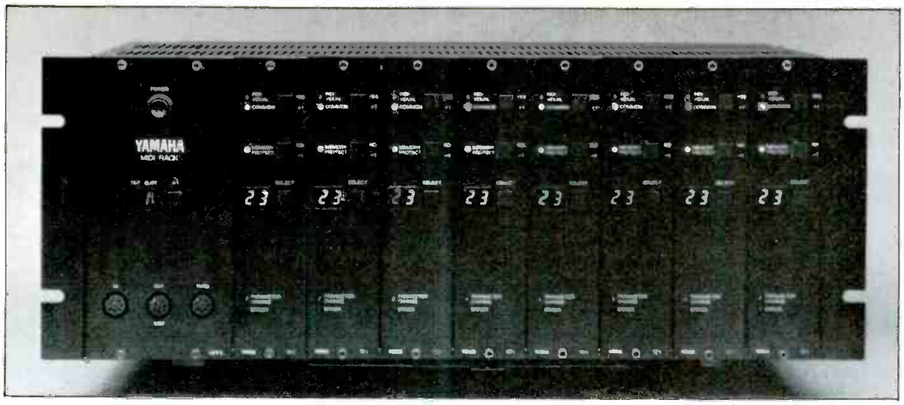
In the keyboard corner, Roland has introduced the JX-8P, a six-voice hybrid digital/analog synthesizer with 64 presets, 32 programmable presets (expandable to 64 via MC-16 cartridge), velocity sensitivity, after touch, and two envelopes per voice. Because of the JX-8P's Amplitude Modulation modes, clangorous bell and percussion sounds, as well as sounds based on complex waveforms, can be created by providing improved cross-modulation possibilities.

Roland also introduced the HP-450 (88-note) and HP-350 (76-note) velocity electric pianos with built speakers, amplifier and MIDI (In, Out, and Thru) and complete MIDI channel assign. The pianos come with 16 different sounds: two pianos, clavinet, harpsichord, vibraphone and electric pianos. There are also on-board chorus, tremolo and brightness controls. Roland also introduced the Axis MIDI remote keyboard, which weighs in at only seven pounds.

Casio made its debut into professional keyboard market. Don't let the size or price of the CZ101 and CZ1000 fool you into thinking anything but serious thoughts about these new synthesizers — Casio has devised a "unique" approach by using phase distortion as a sound source. Separate envelopes are provided for each of the two DCO (digital controlled oscillators), and each envelope can be controlled over eight steps. The CZ101 is an octave keyboard with "Casio-type" keys, while the CZ1000 has an octave keyboard with regular-sized keys.

Yamaha followed through with new technology that supports what is already something of a standard in state-of-the-art synthesizers — the DX-7. The all-new TX-816 is, in reality, eight DX-7s mounted in a rack (something like a rack of Valley People Kepex II noise gates). Well, you may be asking yourself, who needs eight DX-7s? But after hearing only a few of the new sounds that have been programmed especially for the TX-816 rack you'll know the answer. For example, the piano sound pans from left to right, as if you had stereo miked an acoustic piano. Instead of the eight modules playing the same sound detuned (which, incidentally, gives an unbelievable chorus effect), they are pro-

— Roland JX-8P —



— **Yamaha TX-816 Rack Synthesizer** —

grammed to be complementary parts of the harmonic structure that is characteristic of the sound. The eight modules can be used as eight individual DX-7s with separate MIDI channel assignments.

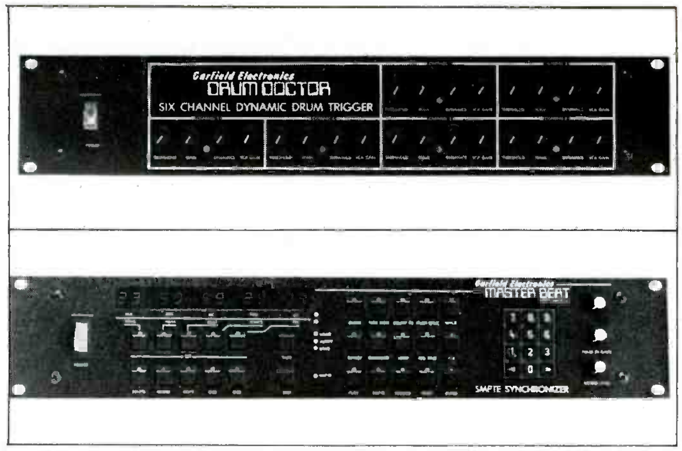
The QX-1, Yamaha's 80,000-note sequencer, was demonstrated with the new TX-816 rack. The unit can memorize sound patches for the entire TX-816 rack or eight individual DX-7s, and all drum programs from either the RX-11 or RX-15 digital drum machines. The QX-1 stores all this data plus the sequence data on an internal disk drive. Eight tracks per bank (sequence) can be assembled via the chain/merge mode into eight different chains (songs). The QX-1 records velocity, after-touch, pitch bend, modulation and patch preset information. After a track has been recorded into the QX-1, it can be edited to the most minute degree: each note of any chord can be examined, and its pitch, velocity, note duration and timing in relation to each measure to the nearest 1/384th of each quarter-note changed. Now that's resolution! The sequencer also records sound program patches from the DX-7, RX-15, RX-11, and TX-816, and stores them to disk. When a particular sequence or chain is played, all the patches will change where they have been programmed to do so.

Yamaha also introduced the first computer dedicated to music composition and production. The CX-5M comes complete with a built-in FM-tone generation system, MIDI In and Out jacks, ASCII keyboard, video monitor port, printer port, joystick controller ports, left and right audio outputs, software cartridge slot, and Microsoft Basic in ROM. Additional software includes the YRM-103 (DX-7 voice editor/-sound storage); YRM-104 Music Marco (which allows the built-in FM sound synth to be accessed through basic programming); YRM-101 FM music composer (which features on-screen graphics of notes); and the YRM-102 (a voicing program for the internal FM synthesizer).

Korg also showed its new DW6000 six-voice, 64-program synthesizer, which features a "Digital Waveform Generator System." Unlike conventional synthesizers, the DW6000 has a selection of eight waveforms for each of the two DCOs. These waveforms are actually digitally recorded waveforms with complex harmonic structures stored on two 56K ROM chips. The DW6000 also features dual digital envelope generators.

J.L. Cooper introduced the Sound Chest II, a digital programable drum unit that features individual modules (up to eight total) in a rack-mounted unit. Each module can hold two sets of sound chips (LinnDrum, Oberheim DMX and Digidrum EPROMs), and each of the chip sets can be modified via the controls on each module, including programmable volume, tuning, dynamic tuning (how far in pitch a sound may be bent by playing dynamics),

— Yamaha QX-1 80,000-note Sequencer —



— Garfield Drum Doctor (*top*) and Master Beat —

MIDI UPDATE

dynamic filter, and decay rate. There is a also a non-programmable pad sensitivity control. After you adjust a module, the settings can then be stored into a master memory module (100-patch capability), along with the settings of the other modules. The Sound Chest also has MIDI In and Out jacks, allowing the patterns to be recorded or played back by any keyboard sequencer. Each module can be assigned a note via MIDI and a MIDI channel or disable MIDI.

Was the"Doctor" in? He sure was, and he had new gadgetry. Of the many **Garfield Electronics** products on show, I only have space here to mention two of them: Master Beat and the Drum Doctor. In essence, Master Beat is a cross between the Doctor Click and Roland's SBX-80 Sync Box. It can generate and read SMPTE timecode, much like the SBX-80, but can also simultaneously generate all other clocks rates (12, 24, 48, 64, 96, plus MIDI Clock and Roland sync). It also has six programmable trigger inputs that can record a live drummer's patterns, and play back a Simmons head, for example.

The Drum Doctor is a six-channel device that converts, for example, live drum tracks on tape into MIDI information. Each channel can be assigned to any MIDI channel and to any note over the entire MIDI range. The device also has a clever feature, in that in can also trigger, for example, a LinnDrum and provide it with velocity information. (A lower-priced version of the Drum Doctor will also be made available, but with less features.)

Software? Did someone say software? Well, there was plenty of software at the NAMM show! Sequencers and voice editors, in particular, were plentiful.

A new **Octave Plateau** 64-track sequencer for an IBM PC equipped with the Roland MPU-401 interface was written for any MIDI-equipped synthesizer. The software can record program changes for each of the 16 MIDI channels, and make multiple program changes within each measure. Tracks can be bounced to other tracks to assemble a song. Filter parameters — such as velocity, note duration, modulation, and pitch bend — can be edited anytime. The program features a graphic display of chords and notes for each measure at a time. You can edit, for example, the pitch of just one note in a complex chord, and see it graphically on screen.

Roland showed the MIDI Processing System (MPS), which comprises software written by Kentyn Reynolds for the IBM PC with MPU-401 MIDI interface. The eight-track sequencer can store up to 60,000 notes in a 640K IBM. Tracks can be bounced and/or copied to other tracks, or appended onto itself, to create songs. MPS has extensive music scoring and editing features; for example, you can write notes on the staves and then hear them played back on the MIDI synth of your choice. Scores can be transposed graphically on staves and printed on any IBM-compatible printer with graphics. Roland also showed the MUSE written by Jeff Rona for the Apple IIe with MPU-401. MUSE is rather like a on-screen MSQ-700 for the Apple — as with the MSQ, MUSE has eight tracks with chain mode and step editing capabilities. A game joystick can be used as sort of a mouse to control record/play mode and all other functions.

Roger Powell, a member of the band Utopia, also has written a program based around Roland's MPU-401 interface. Distributed by Cherry Lane Technologies, Texture has been written for both Apple IIe and IBM PC microcomputers, and features eight tracks per sequence, 16 sequences total. You can bounce tracks, edit velocity parameters, display MIDI data, and even offset the feel of each track against each other.

Cherry Lane Technologies also showed the JMS 12-track sequencer for use on a Commodore 64. The JMS system, which requires its own interface, is made up of five different modules: a sequencer, song merger, arpeggiator, a DX-7 editor, and a DX-7 voice manager. The company also showed Connections, an eight-track sequencer for the Apple IIe; DX-Heaven, a DX-7 voice editor and sound-storage program for the Apple IIe; and CZ-Rider, a voice editor program for the Apple IIe plus Casio CZ-101 and CZ-1000 synthesizers.

Mimetics Corporation showed yet another DX-7 voice editor, called Data-7, and a voice-storage program, Performance-7, which can store, recall and swap 288 voices instantly. Both programs are available in versions for the Commodore 64 with the Yamaha MIDI interface card; an Apple IIe with either the Yamaha MIDI interface or Roland's MPU-401; and an IBM with the MPU-401.

All in all, the Winter NAMM exhibition was a great show and, for those that didn't get the chance to be there, I hope this article will serve to satisfy your MIDI curiosity. ■■■

The Directory

R-e/p's Product Listing of RECORDING, PRODUCTION, and LIVE-PERFORMANCE CONSOLES

ACES
U.S. Distributor:
Mammoth Marketing
P.O. Box 6493
Thousand Oaks, CA 91359
Phone: (805) 496-2969

MC-24 Recording Console
Inputs: 28 to 52.
Bus Outs: 24.
Monitor Selection: In-line I/O type.
Auxiliary Sends: Eight.
EQ Section: Three-band, fully parametric; two-band shelving, variable hi and low pass filters.
Metering: VU/LED array (optional).
Automation Capability: Fader level, mute, routing.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: RMS compressor/limiter and nine-segment, three-color LED array on each input; AudioFad conductive plastic faders; stand.
Price Range: Automation-ready: $32,250 to $56,850; with automation: $42,550 to $75,950.

SM16 MkII
Inputs: 24 to 36.
Bus Outs: 16.
Monitor Selection: Split-type; up to 32-track.
Auxiliary Sends: Five.
EQ Section: Five-band/three-mid switchable.
Metering: VU/LED array (optional).
Automation Capability: N/A.
Frequency Response: 40 Hz to 20 kHz, ±3 dB.
Distortion: 0.05% at 1 kHz reference.
Signal-to-Noise Ratio: –125 dB EIN.
Selected Standard Features: Nine-segment, three-color LED array on each input; integrated wired patchbay; 16 auxiliary returns-bus assigns; Audiofad conductive-plastic faders optional.
Price Range: $11,400 to $17,150.

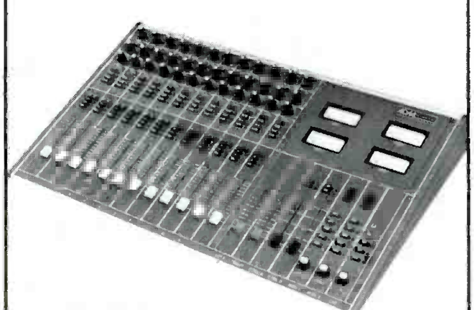
ML24 MkII
Inputs: 28 to 56.
Bus Outs: 24.
Monitor Selection: I/O type.
Auxiliary Sends: Five.
EQ Section: Five-band/three-mid switchable.
Metering: VU/LED array (optional).
Automation Capability: N/A.
Frequency Response: 30 Hz to 20 kHz, ±3 dB.
Distortion: 0.05% at 1 kHz reference.
Signal-to-Noise Ratio: –125 dB EIN.
Selected Standard Features: Nine-segment, three-color LED array on each input; integrated wired patchbay.
Price Range: $15,350 to $30,350.

SM24 MkII
Inputs: 28 to 36.
Bus Outs: 24.
Monitor Selection: Split-type; up to 48-track.
Auxiliary Sends: Five.
EQ Section: Five-band/switchable mids.
Metering: VU/LED (optional).
Automation Capability: N/A.
Frequency Response: 40 Hz to 20 kHz, ±3 dB.
Distortion: 0.05% at 1 kHz reference.
Signal-to-Noise Ratio: –125 dB EIN.
Selected Standard Features: Three-color, nine-segment LED array; integrated wired patchbay; 24 auxiliary return/bus assigns; Audiofad conductive-plastic faders optional.
Price Range: $15,800 to $19,650.

MM-16
Inputs: 40.
Bus Outs: 16 × two.
Monitor Selection: N/A.
Auxiliary Sends: N/A.
EQ Section: Five-band/three switchable mids.
Metering: VU/LED array (optional).
Automation Capability: N/A.
Frequency Response: 40 Hz to 20 kHz, ±3 dB.
Distortion: 0.05% at 1 kHz reference.
Signal-to-Noise Ratio: –125 dB EIN.
Selected Standard Features: 16 mixes plus stereo out; integrated wired patchbay; nine-segment, three-color input LED array on each input; Audiofad conductive-plastic faders optional.
Price Range: $18,750.

ADM TECHNOLOGY, INC.
1626 East Big Beaver Road
Troy, MI 48063
Phone: (313) 524-2100

Post-Pro
Inputs: 12.
Bus Outs: Two.
Monitor Selection: Two monitor busses.
Auxiliary Sends: N/A.



EQ Section: Three-band with high/low pass filter.
Metering: Output.
Automation Capability: Parallel video editor interface will control the console's two VCA group A-B bussing system.
Frequency Response: With no EQ: ±1 dB, 20 Hz to 20 kHz at 1 kHz reference.
Distortion: THD at +24 dBm or lower will be less than 0.07% and will not exceed 0.15%, 100 Hz to 20 kHz.
Signal-to-Noise Ratio: 20 Hz to 20 kHz, –80 dB.
Selected Standard Features: 12 adjustable line level inputs; VCA slidex attenuators; full or partial auto/manual control; dual monitor busses; video editor interface; two output VCA control groups.
Price Range: $15,260.

ALLEN & HEATH BRENELL, LTD.
Five Connair Road
Orange, CN 06477
Phone: (203) 795-3594

SR-8/16/24/416/424
Inputs: Eight/16/24/16/24.
Bus Outs: L/R plus mono; 416/424 : four plus L/R mono.

Monitor Selection: Headphone; second control room.
Auxiliary Sends: Four.
EQ Section: Four-band.
Metering: Two LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: Typically less than 0.05% THD.
Signal-to-Noise Ratio: –86 dB.
Selected Standard Features: External PS; 48 volt phantom power standard.
Price Range: SR-8: $1,100; SR-16: $1,800; SR-24: $2,500; SR-416: $2,500; SR-424: $3,200

SRM-186/248
Inputs: 18 plus two/24 plus two.
Bus Outs: Six.
Monitor Selection: "Comprehensive engineering."
Auxiliary Sends: N/A.
EQ Section: Three-band with sweep mid.
Metering: Seven LED/nine LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: Typically less than 0.05% THD.
Signal-to-Noise Ratio: EIN –125 dB.

first the bad news

Your old fashioned 24-track console is obsolete. It was designed for music recording, but that's only the beginning for contemporary multipurpose studios. Now you've got synthesizer dates needing MIDI interface, audio for video with computer editor control. Throw in a mix minus requirement and a few stereo input lines that need EQ. Put compression on a subgroup and then subgroup it again. The producer needs a rough in ten minutes and you're down to your last patchcord. Bad news.

and now the good news

The new **Elite** consoles are a step beyond anything you've ever seen. They've been designed from the ground up to solve the problems you'll face tomorrow. For starters, they're 26-track consoles. Each input group has two separate channels, and all major functions assign to either fader or monitor path. The Elite operating system sets new standards of flexibility, yet it is easily understood. There are several automation options, and logic systems offer direct digital interface. Best of all, new circuits and a refined parametric equalizer deliver that unbeatable NEOTEK sound.

Get ready to change the way you think about multitrack recording.

it's time to meet the

ELITE

anything less is yesterday's news

NEOTEK CORP

1154 West Belmont Avenue

Chicago, Illinois 60657 U.S.A. 312-929-6699

The Directory

Selected Standard Features: On-stage monitoring mixer with built-in splitter system; road case included.
Price Range: SRM-186: $3,600; SRM-248: $4,800

System 8-168D/1616D/2416D
Inputs: 16/24.
Bus Outs: Eight plus L/R.
Monitor Selection: Eight/16 channel.
Auxiliary Sends: Three, plus stereo cue.
EQ Section: Three-band.
Metering: 10 mechanical.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: Mike input to output at +4 dBv, 45 dB gain, less than 0.05%, THD 20 Hz to 20 kHz.
Signal-to-Noise Ratio: EIA –125 dB/+86 dB for outputs.
Selected Standard Features: All steel construction; switchable outputs between +4 and –8 operation.
Price Range: 8-16D: $4,200; 1616D: $4,800; 2416D: $5,900.

System 8-EX8
Inputs: Eight.
Bus Outs: By tie lines.
Monitor Selection: N/A.
Auxiliary Sends: Three.
EQ Section: Three-band.
Metering: N/A.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: Mike input to output at +4 dBv, 45 dB gain less than 0.05%; THD 20 Hz to 20 kHz.
Signal-to-Noise Ratio: EIA –125 dB/+86 dB group outputs.
Selected Standard Features: Expander unit for System 8 Series; can be used as standalone mixer without masters.
Price Range: $1,800

CMC-16/24/32
Inputs: Eight (24 at mixdown); 16 (40 at mixdown); and 24 (56 at mixdown), respectively.
Bus Outs: Eight, 16, and 24 (all plus mono L/R) respectively.
Monitor Selection: Eight, 16, and 24 respectively.
Auxiliary Sends: Six.
EQ Section: Three-band.
Metering: N/A.
Automation Capability: Bus assignment and muting.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: Mike input to output at +4 dBv, 45 dB gain less than 0.05%; THD 20 Hz to 20 kHz.
Signal-to-Noise Ratio: –86 dB group outputs.
Selected Standard Features: I/O design; Alps faders as standard.
Price Range: Eight-channel: $4,100; 16-channel: $5,600; 24-channel: $7,300

Syncon-B
Inputs: Eight to 44.
Bus Outs: Eight to 24 plus L/R.
Monitor Selection: Eight to 44.
Auxiliary Sends: Four.
EQ Section: Four-band plus variable low-cut filter.
Metering: LEDs 10 to 46.
Automation Capability: Automation-ready.
Frequency Response: 20 Hz to 20 kHz, ±0.5 dB.
Distortion: 0.005% at 1 kHz, THD IMD less than 0.015% SMPTE.
Signal-to-Noise Ratio: Channel output noise, line input routed at zero dB gain, –1 dB reference 0VU.
Selected Standard Features: Expandable configurations; PPG channel faders standard.
Price Range: $9,980 (8 in/out) to $39,980 (44 in/24 out)

AKAI
U.S Distributor:
International Music Company
1316 East Lancaster
Fort Worth, TX 76102
Phone: (817) 336-5114

MG1212 Mixer/Recorder
Inputs: 12.
Bus Outs: 12 × two.
Monitor Selection: Stereo.
Auxiliary Sends: Four.
EQ Section: Three-band parametric.
Metering: LED ladder.
Automation Capability: Auto location, punch in/out, playback mute.
Frequency Response: 40 Hz to 20 kHz.
Distortion: 0.5%.
Signal-to-Noise Ratio: –94 dB NAB A weighted.
Selected Standard Features: Digital assigns; built-in 14-track recorder using custom half-inch cassette format; auto punch in/out; sync track.
Price Range: $6,995

AMEK CONSOLES, INC.
10815 Burbank Boulevard
North Hollywood, CA 91601
Phone: (818) 508-9788

Scorpion
Inputs: 16 to 32.
Bus Outs: Eight or 16.
Monitor Selection: Separate stereo.
Auxiliary Sends: Four.
EQ Section: Four-band, sweep mids.
Metering: LED, PPM/VU balance.
Automation Capability: Optimix I.
Frequency Response: 7 Hz to 30 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: 16 bus; scaled down version of Matchless.
Price Range: $5,500 to $13,500.

TAC Matchless
Inputs: 16 to 26.
Bus Outs: 24.
Monitor Selection: In-line stereo.
Auxiliary Sends: Eight.
EQ Section: Eight shelving, plus two sweep mids.
Metering: LED, PPM/VU balance.
Automation Capability: Optimix I.
Frequency Response: 7 Hz to 29 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Separate mike and line gain; eight sends; eight returns; two mute busses; eight mixing subgroups; separate monitor bus which attaches to two mixing busses for a total of 60 line sources in the stereo busses.
Price Range: $20,000 to $25,000.

BCD1 Series II
Inputs: 6 to 16.
Bus Outs: Two or four.
Monitor Selection: Yes.
Auxiliary Sends: Two or four.
EQ Section: Three-band.
Metering: PPM or VU.
Automation Capability: Optional VCAs, Audio Kinetics MasterMix, Massenberg Moving Fader Automation.
Frequency Response: 7 Hz to 33 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Small portable; field engineered for stereo inputs; Audio follows Video; serial or parallel.
Price Range: $6,000 to $16,000.

M1000 Series II
Inputs: Eight to 48.
Bus Outs: Two, four, eight, or 10.
Monitor Selection: Separate or in-line stereo.
Auxiliary Sends: Four or eight.
EQ Section: Three-band selectable; four-band parametric.
Metering: VU or LED.
Automation Capability: VCA, Audio follows Video, serial or parallel.
Frequency Response: 7 Hz to 33 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Multi-section of modules from resettable routing; simple EQ and monitoring to full parametric EQ plus complex monitoring.
Price Range: $15,000 to $50,000.

M2500 STP (Stereo Teleproductions)
Inputs: 20 to 56.
Bus Outs: 24.
Monitor Selection: In-line stereo.
Auxiliary Sends: Six or eight.
EQ Section: Four-band parametric.
Metering: VU or LED.
Automation Capability: Massenberg, Mastermix, ARMS.
Frequency Response: 7 Hz to 39 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Master status switching; patching to over 1,000 points; 48 metering and monitoring; stereo line inputs.
Price Range: $55,000 to $140,000.

M3500
Inputs: 36 to 56.
Bus Outs: 24 or 48.
Monitor Selection: In-line, 24 to 48.
Auxiliary Sends: Eight.
EQ Section: Four-band fully parametric.

The Directory

Metering: VU or LED.
Automation Capability: Massenberg, Mastermix, Sound Workshop ARMS.
Frequency Response: 7 Hz to 39 kHz, ±0.5 dB.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Master status switching; monitor section has 60mm fader; resettable sends and routing and patching to 2,000 points.
Price Range: $150,000 to $450,000.

Models 4000-4600

Inputs: 36 to 120.
Bus Outs: 24, 32, or 48.
Monitor Selection: In-line or separate.
Auxiliary Sends: Two, four or eight.
EQ Section: Four-band fully parametric, with sweep hi/low pass filters.
Metering: VU and LED.
Automation Capability: Massenberg, Mastermix, ARMS.
Frequency Response: 7 Hz to 39 kHz, ±0.5 dB.
Distortion: 0.002%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Three- or four-man section consoles; four, six or eight outputs; 24 track routing.
Price Range: $200,000 to $650,000.

M6500 STP

Inputs: 32 to 96.
Bus Outs: 32 or 48.
Monitor Selection: In-line.
Auxiliary Sends: Eight.
EQ Section: Fully parametric EQ.
Metering: VU and LED.
Automation Capability: Mastermix or Massenberg.
Frequency Response: 7 Hz to 39 kHz.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Digital assignments to all busses from all input sources; completely resettable 16 memories on-board; complete dynamics on all inputs; five stereo outputs; two mike and two line inputs per I/O.
Price Range: $180,000 to $500,000.

Angela

Inputs: 16 to 56.
Bus Outs: 24.
Monitor Selection: In-line stereo.
Auxiliary Sends: Six.
EQ Section: Eight fixed shelving, plus two full sweep.
Metering: LED PPM, and VU.
Automation Capability: Mastermix, ARMS, Optimix.
Frequency Response: 7 Hz to 39.5 kHz.
Distortion: 0.002%.
Signal-to-Noise Ratio: Better than –90 dB at 200 Hz.
Selected Standard Features: Separate mike/line trim; four-band EQ; four stereo subgroups; patching to over 800 points; all hand wired.
Price Range: $25,000 to $65,000.

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The Directory

AUDIOARTS ENGINEERING
5 Collins Road
Bethany, CT 06525
Phone: (203) 393-0887

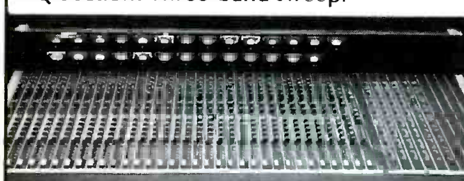
MTX-80

Inputs: 16 to 48
Bus Outs: Eight plus matrix.
Monitor Selection: N/A.
Auxiliary Sends: Four (selectable pre-post).
EQ Section: Three-band sweep with HPF.
Metering: Sends, groups, master left and right.

Automation Capability: N/A.
Frequency Response: Overall In/Out 20 Hz to 20 kHz, ±0.5 dB for line; 30 Hz to 20 kHz ±0.5 dB for mike.
Distortion: Line 20 Hz to 20 kHz at +18 dB 0.02%.
Signal-to-Noise Ratio: 20 kHz bandwidth −85 dBm.
Selected Standard Features: Phantom power; pre/post send select; M-104 conductive-plastic fader; 11 by one matrix on each group; totally modular construction.
Price Range: $13,500 to $28,000.

Model 3224

Inputs: 24 to 32
Bus Outs: 24.
Monitor Selection: Monitor and cue.
Auxiliary Sends: Four.
EQ Section: Three-band sweep.



Metering: 24-track.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: 20 Hz to 20 kHz at +20 dBm, 0.05%.
Signal-to-Noise Ratio: −129 EIN, 20 kHz.
Selected Standard Features: I/O design; totally modular construction; comprehensive solo system; phantom power; M104 conductive-plastic faders; control module with talkback, slate, and modulator.
Price Range: $19,500 to $23,900.

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

Model UA800

Inputs: Up to 64.
Bus Outs: 32.
Monitor Selection: Up to 64.
Auxiliary Sends: 16.

EQ Section: Four-band parametric.
Metering: Bargraph PPM/VU switchable.
Automation Capability: Audio Kinetics Master Mix system.
Frequency Response: +0/−0.25 dB 20 Hz to 20 kHz.
Distortion: 30 Hz at +10 dB is 0.037%; 1 kHz at +10 dB is 0.017%; 20 kHz at 0 dB is 0.035%; 20 kHz at +10 dB is 0.055%. (Note: Second harmonic only.)
Signal-to-Noise Ratio: −82.5 dB (20 Hz to 20 kHz unweighted).
Selected Standard Features: Signal processing on a per channel basis: compressor/peak limiter/expander/gate plus equalizer.
Price Range: $195,000 for 48-input console.

M Series

Inputs: Up to 24.
Bus Outs: Two plus mono sum.
Monitor Selection: Main/aux/play.
Auxiliary Sends: Four.

EQ Section: Three-band plus HF and LF.
Metering: VU or PPM analog meters.
Automation Capability: N/A.
Frequency Response: 30 Hz to 20 kHz, ±0.5 dB; 18 Hz to 30 kHz, −3 dB and falling due to steep cut filters.
Distortion: At +12 dBu: better than 0.05% at 30 Hz to 16 kHz.
Signal-to-Noise Ratio: Better than −80 dB at 20 Hz to 20 kHz.
Selected Standard Features: Optional multi-ratio compressor/limiter modules.
Price Range: $9,000 for 8 channel to $20,000 for 24 channel.

AUDIO PROCESSING SYSTEMS
90 Oak Street
Newton Upper Falls, MA 02164
(617) 965-1200

Model 3000

Inputs: 16 to 56.
Bus Outs: 24-track, stereo.
Monitor Section: Stereo control room; two cue studio.
Auxiliary Sends: Four to eight mono; one stereo.
EQ Section: Four-band, semi-parameteric with shelving.
Metering: 12-segment LED, VU, or PPM.
Automation Capability: Sound Workshop ARMS; Audio Kinetics MasterMix.
Frequency Response: 20 Hz to 20 kHz, −0.5 dB.
Distortion: 0.01% typical; 0.1% (0.2% with automation) maximum.
Signal-to-Noise Ratio: Better than −96 dB (−92 with automation) maximum.
Selected Standard Features: Status control; four mixdown modes; totally modular; expansion and retrofit capable.
Price Range: $20,000 to $70,000

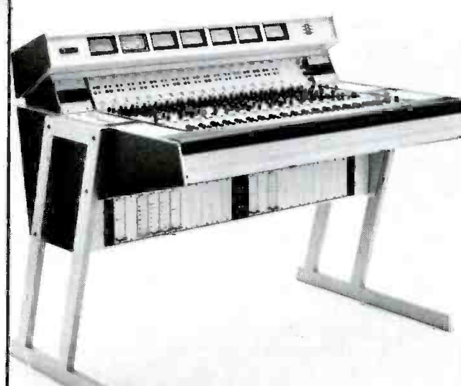
Model 2200

Inputs: Up to 56.
Bus Outs: Omni-bus system 12- to 24-track; may be cross matrixed.
Monitor Section: "Options available."
Auxiliary Sends: N/A.
EQ Section: Four-band, semi-parameteric with shelving.
Metering: 12-segment LED, VU, or PPM.
Automation Capability: Sound Workshop ARMS.
Frequency Response: 20 Hz to 20 kHz, −0.5 dB.
Distortion: 0.01% typical; 0.1% (0.2% with automation) maximum.
Signal-to-Noise Ratio: Better than −91 dB (−89 with automation) maximum.
Selected Standard Features: Rugged; operator controlled matrixing and sequencing; requires two module types.
Price Range: $20,000 to $70,000

AUDITRONICS
3750 Getwell Road
Memphis, TN 38118
Phone: (901) 362-1350

300 Series

Inputs: Mono to stereo to 32.
Bus Outs: Four or eight.
Monitor Selection: Monitor-mix selection.



Auxiliary Sends: Four.
EQ Section: Optional three-band plus HP/LP.
Metering: VU or PPM.
Automation Capability: N/A
Frequency Response: 20 Hz to 20 kHz, –0.5 dB.
Distortion: Less than 0.05% THD 20 kHz, +24 dBv.
Signal-to-Noise Ratio: Better than –84 dB.
Selected Standard Features: VCA control; submastering; user-programmable logic control system.
Price Range: $20,000 to $50,000

700 Series

Inputs: 48.
Bus Outs: 16 or 48.
Monitor Selection: In-line.
Auxiliary Sends: Six.
EQ Section: Three-band HP/LP.
Metering: VU or PPM.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, –1 dB.
Distortion: Less than 0.01%; THD 20 kHz, +24 dBv.
Signal-to-Noise Ratio: Better than –80 dB.
Selected Standard Features: Special mainframe versions specifically for mobile teleproductions vehicles.
Price Range: From $40,000

AUDIO TECHNICA U.S., INC.
1221 Commerce Drive
Stow, OH 44224
Phone: (216) 686-2600

AT RMX64 Mixer/Recorder

Inputs: Six.
Bus Outs: Four.
Monitor Selection: Headphone monitor.
Auxiliary Sends: Two pre/post fader.
EQ Section: Two-band, quasi-parametric.
Metering: Four VUs.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1.5 dB.
Distortion: 0.05% THD.
Signal-to-Noise Ratio: Better than –122 dBv EIN.
Selected Standard Features: Built-in four-track, 3¾ ips cassette deck (stereo at 1⅞ ips) with selectable Dolby B/C noise reduction; soloing, phantom power.
Price Range: $1,495.

BIAMP SYSTEMS, INC.
P.O. Box 728
Beaverton, OR 97075
Phone: (503) 641-6767

83B Series

Inputs: Six to 24.
Bus Outs: Two submasters and main.
Monitor Selection: Pre-EQ on each channel.
Auxiliary Sends: One effects, post fader.
EQ Section: HF = ±18 dB at 8 kHz (shelving); MF = ±15 dB at 2.5 kHz (peaking); LF = ±18 dB at 50 Hz (shelving).
Metering: LED ladders.
Automation Capability: N/A.
Frequency Response: ±0.5 dB, 20 Hz to 20 kHz.
Distortion: THD 0.02% maximum; 42 dB gain, +8 dBm out.
Signal-to-Noise Ratio: EIN –129 dBv; maximum output +18 dBm; residual noise is –96 dBv.
Selected Standard Features: Discrete bipolar input circuitry; available in eight, 12-, 16-, and 24-channel consoles, as well as six and eight rackmount versions.
Price Range: $799 to $1,499

24 and 28 Series

Inputs: Eight to 32/12 to 32, respectively.
Bus Outs: Four submasters/eight submasters.

Monitor Selection: Stereo/dual monitor outputs with solo headphone, record/live functions.
Auxiliary Sends: Two effects, pre/post fader.
EQ Section: HF = ±15 dB at 8 kHz (shelving); MF = ±12 dB at 140 Hz to 8.5 kHz (peaking); LF = ±15 dB at 100 Hz (shelving).
Metering: 14-segment florescent displays.
Automation Capability: N/A.
Frequency Response: –0.5 dB, 20 Hz to 20 kHz.
Distortion: THD 0.04% maximum; 40 dB gain, +4 dBm out.
Signal-to-Noise Ratio: EIN –128 dBv (unweighted); maximum output is +22 dBm; residual noise is –94 dBv.
Selected Standard Features: Discrete bipolar input circuitry; patch insert points on channels and submasters; direct outs and tape returns on all input channels; channel monitors selectable from pre-EQ/post fader/tape return; main and monitor selectable from mix/submasters/channel monitors.
Price Range: 24 Series: $1,499 to $6,299; 28 Series: $2,799 to $6,899

Bimix Series

Inputs: Eight to 32.
Bus Outs: Four to 24 submasters.
Monitor Selection: Selectable sources at control room, studio, and headphone, includes solo, talkback, slate oscillator.
Auxiliary Sends: Two effects plus two cue.
EQ Section: Three-band sweepable.

Metering: LED ladders.
Automation Capability: N/A.
Frequency Response: –0.5 dB, 20 Hz to 20

The Directory

kHz.
Distortion: THD 0.04% maximum; 40 dB gain, +4 dBm out.
Signal-to-Noise Ratio: EIN −128 dBv (unweighted); maximum output is +22 dBm; residual noise −94 dBv.
Selected Standard Features: Signal and metering sensitivity adjustable +4 to −10; expandable input/output module design; discrete bi-polar input circuitry; patch insertion points on all channels and submasters.
Price Range: $3,999 to $12,099

CARVIN MANUFACTURING CORP.
1155 Industrial Avenue
Escondido, CA 92025
Phone: (619) 747-1710

MX-1688
Inputs: 16.
Bus Outs: Eight.
Monitor Selection: Pre/post switchable, two-band, cue out.
Auxiliary Sends: Two can be used as effects or monitor sends.
EQ Section: Three-band, fully sweepable, parametric with switchable in/out.
Metering: All subgroups outs/monitors-two-tracks.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz.
Distortion: 0.005%.
Signal-to-Noise Ratio: Better than −90 dB.
Selected Standard Features: Complete eight- to two-track mixdown section with stackable inputs.
Price Range: $2,295.

COHERENT COMMUNICATIONS
13756 Glenoaks Boulevard
Sylmar, CA 9132
Phone: (818) 362-2566

Model MX-80
Inputs: Four.
Bus Outs: Two.
Monitor Selection: One O/P with gain control.
Auxiliary Sends: N/A.
EQ Section: High pass each I/O mid frequency.
Metering: VU plus peak LED.
Automation Capability: N/A.
Frequency Response: −1 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.1%, 50 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than −80 dB.
Selected Standard Features: Mini-mixer, batttery powered for film and video.
Price Range: $1,900

CONNECTRONICS CORP.
652 Glenbrook Road
Stamford, CT 06906
Phone: (800) 322-2537

Model 62/122
Inputs: six/12.
Bus Outs: two/two.
Monitor Selection: N/A.
Auxiliary Sends: Four/four.
EQ Section: High shelving 11 kHz; mid sweep; 3 to 6.5 kHz; 45 Hz.
Metering: Two × 12 section LED bargraph on/off peak hold.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, −2 dB.
Distortion: N/A.
Signal-to-Noise Ratio: Maximum all outputs at −84 dB.
Selected Standard Features: Portable; balanced inputs.
Price Range: $1,345 to $1,995.

CREST AUDIO
150 Florence Avenue
Hawthorne, CA 07506
Phone: (213) 423-1300

Ten Series
Inputs: Eight, 12, 16, and 24.
Bus Outs: L/R mono.
Monitor Selection: One monitor send.
Auxiliary Sends: One pre; two post.
EQ Section: Four-band fixed.
Metering: Two LED arrays.
Automation Capability: N/A.
Frequency Response: ±0.5 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.3%.
Signal-to-Noise Ratio: Better than −119 dB.
Selected Standard Features: 48V phantom power; mike/line switch and peak indicator on each channel; built-in reverb; reverb and effects to monitor switches; complete solo system; two stereo returns.
Price Range: N/A

Twenty and Forty Series
Inputs: Eight to 32.
Bus Outs: Four groups, L/R mono.
Monitor Selection: In-line tape monitoring with monitor send.
Auxiliary Sends: Four.
EQ Section: High and low selectable center frequency; mid, continuously variable center frequency.
Automation Capability: N/A.
Frequency Response: ±0.5 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.3%.
Signal-to-Noise Ratio: Better than −119 dB.
Selected Standard Features: Fully modular and line gain control; 20-dB pad; selectable auxiliary send; channel muting; six sub mono configuration available; complete communications; all sends and returns soloable; Forty Series has eight subgroups.
Price Range: N/A

Sixty and Eighty Series
Inputs: 16 to 32.
Bus Outs: Eight, plus L/R mono.
Monitor Selection: In-line tape monitoring with monitor sends.
Auxiliary Sends: Six.
EQ Section: Three-band sweepable.
Metering: 10 LED arrays.
Automation Capability: N/A.
Frequency Response: ±0.5 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.3%.
Signal-to-Noise Ratio: Better than −119 dB.
Selected Standard Features: Fully modular, 48V phantom; separate mike/line gain controls; EQ out switch; line-up oscillator; conductive-plastic pots used throughout; all send and returns soloable; Eighty Series features 16 subgroups.
Price Range: N/A

Seventy Series
Inputs: 16 to 32.
Bus Outs: 16 to 32
Monitor Selection: N/A.
Auxiliary Sends: Four.
EQ Section: Four-band, fully parametric.
Metering: 10 LED arrays.
Automation Capability: N/A.
Frequency Response: ±0.5 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.3%.
Signal-to-Noise Ratio: Better than −119 dB.
Selected Standard Features: Fully modular, 10 bus stage monitoring version of Sixty Series.
Price Range: N/A

FOSTEX
15431 Blackburn Avenue
Norwalk, CA 90650
Phone: (213) 921-1112

Model 250 Mixer/Recorder
Inputs: Four mike and two auxiliary.
Bus Outs: Monitor mix, monitor auxiliary, two buss left/right; and four directs.
Monitor Selection: Four input/mono out.

Auxiliary Sends: Four send/mono out.
EQ Section: ±12 dB at 4 kHz, and 100 Hz.
Metering: Four-channel bus.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: THD 0.05% at 1 kHz nominal level.
Signal-to-Noise Ratio: Overall −75 dB weighted.
Selected Standard Features: Built-in, four-track cassette recorder; headphone amplifier. Dolby C noise reduction.
Price Range: $1,300.

Model 350
Inputs: Eight line/mike and four auxiliary.
Bus Outs: Four buss, eight directs, and two auxiliary.
Monitor Selection: Eight in/two out.
Auxiliary Sends: Two.
EQ Section: ±12 dB two sections; 800 Hz to 12 kHz; 80 Hz to 1.2 kHz.
Metering: Four buss/two auxiliary.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: THD 0.03% at 1 kHz nominal level.
Signal-to-Noise Ratio: One mike −68 dB weighted, eight mike −58 dB weighted; one line −75 dB weighted, eight line −65 dB weighted.

The Directory

Selected Standard Features: Two stereo phono pre-amps; built-in stereo headphone amp; monitor select; grouping in mixdown.
Price Range: $1,120.

GOTHAM AUDIO/EMT FRANZ
741 Washington Street
New York, NY 10014
Phone: (212) 741-7411

EMT 10.08.2
Inputs: 10 to 30.
Bus Outs: Two to 18.
Monitor Selection: Stereo.
Auxiliary Sends: Two.
EQ Section: High/mid/low, switchable.
Metering: Peak and correlation.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz.
Distortion: Maximum 0.1%, minimum 80 dB with 14 dB headroom.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Available in portable or fixed configurations; "flexible output modules for recording and sound reinforcement applications."
Price Range: $12,750 to $49,795.

GML, INC.
2323 Corinth Avenue
West Los Angeles, CA 90064
Phone: (213) 479-7471

Model 7900
Inputs: 48 to 72.
Bus Outs: 24.
Monitor Selection: Separate 24 × six.
Auxiliary Sends: Eight.
EQ Section: Three-band parametric.
Metering: 10-segment LED.
Automation Capability: Moving faders and switches; disk-based system.
Frequency Response: 3 Hz to 200 kHz before rolloff.
Distortion: 0.006% IM at +20 dBv.
Signal-to-Noise Ratio: Line in/–85 dBv.
Selected Standard Features: N/A.
Price Range: $250,000 to $350,000.

HARRISON SYSTEMS
P.O Box 22964
Nashville, TN 37202
Phone: (615) 834-1184

MR2
Inputs: 60.
Bus Outs: 48.
Monitor Selection: In-line.
Auxiliary Sends: Eight mono plus one stereo.
EQ Section: Four-band, fully parametric.
Metering: 36-segment bargraph VU/PPM.
Automation Capability: Audio Kinetics Mastermix, or Harrison Auto Set 1.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Single button status select; Sigma cue function; separate power supply.
Price Range: $91,000 to $166,000

Series 3
Inputs: Up to 56.
Bus Outs: 24.
Monitor Selection: In-line.
Auxiliary Sends: Four mono plus one stereo.
EQ Section: Three-band, fully parametric.
Metering: 36-segment bargraph VU/PPM.
Automation Capability: MasterMix or Auto Set 1.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Single-button status select; Sigma cue function.
Price Range: $61,000 to $124,000

Series 4
Inputs: Up to 60.
Bus Outs: Up to 24.
Monitor Selection: In-line or groups.
Auxiliary Sends: Four.
EQ Section: Three-band, sweep.
Metering: 36-segment bargraph LED.
Automation Capability: Harrison Auto Set 1 or Mastermix.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Center dentented EQ and pan control; console mounted optional BTX Soft touch timecode synchronization system.
Price Range: $35,000 to $100,000

Series 5 Sound Reinforcement
Inputs: 32 with extender capability.
Bus Outs: 33.
Monitor Selection: Monitor any point.
Auxiliary Sends: 16 plus 16.
EQ Section: Four-band, fully paramteric.
Metering: 16 each bargraph LED.
Automation Capability: N/A.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Stage or house monitoring sound reinforcement console.
Price Range: $60,000 to $115,000

Series 7
Inputs: Up to 28
Bus Outs: Two stereo pairs plus two mono.
Monitor Selection: Stereo.
Auxiliary Sends: Up to two.
EQ Section: Three-band sweep.
Metering: Moving-coil VU or bargraph PPM.
Automation Capability: MasterMix, Sound Workshop ARMS, or Harrison VSI.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Broadcast or small production console.
Price Range: $10,000 to $40,000

Raven
Inputs: 28 to 36.
Bus Outs: 24 plus direct.
Monitor Selection: In-line.
Auxiliary Sends: Four.
EQ Section: Three-band, sweep.
Metering: Moving coil VU.
Automation Capability: Mastermix, ARMS, or Harrison VSI.
Frequency Response: N/A.
Distortion: N/A.
Signal-to-Noise Ratio: N/A.
Selected Standard Features: Low-cost, multitrack console.
Price Range: $35,000 to $43,000

HILL AUDIO, INC.
231 Marquis Court
Lilburn, GA 30247
Phone: (404) 923-3193

MULTIMIX
Inputs: 16.
Bus Outs: Four.
Monitor Selection: Four to eight.
Auxiliary Sends: Two.
EQ Section: Three-band, fixed.
Metering: LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±2 dB.
Distortion: Less than 0.04% THD; less than 0.02% IMD.
Signal-to-Noise Ratio: –126 dB.
Selected Standard Features: Rack mount; phantom power on all inputs.
Price Range: $1,799.

Model B3
Inputs: 16 to 32.
Bus Outs: Four to eight.
Monitor Selection: Four to eight.
Auxiliary Sends: Three.
EQ Section: Four-band; two sweep.
Metering: VU plus LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±0.2 dB.
Distortion: Less than 0.02% THD; less than 0.04% IMD.
Signal-to-Noise Ratio: –126.
Selected Standard Features: Independent line return section; monitor section; modular construction; external PSU.
Price Range: $3,000 to $8,000.

Model J3
Inputs: 16 to 48.
Bus Outs: Eight.
Monitor Selection: Eight to 16.
Auxiliary Sends: Six.
EQ Section: Eight-band or 24 point.
Metering: LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±0.2 dB.
Distortion: Less than 0.008% THD; less than 0.01% IMD.
Signal-to-Noise Ratio: –126 dB.
Selected Standard Features: Phase-coherent EQ section; modular construction; external PSU; 48 ph.p.
Price Range: $8,000 to $24,000.

STAGE MIX
Inputs: 12.
Bus Outs: Six.
Monitor Selection: N/A.
Auxiliary Sends: N/A.
EQ Section: Three- and four-band.
Metering: LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±2 dB.
Distortion: Less than 0.05% THD; less than 0.08% IMD.
Signal-to-Noise Ratio: –126 dB.
Selected Standard Features: Rack-mount monitor console; transformer splitter on inputs; tranformer outputs.

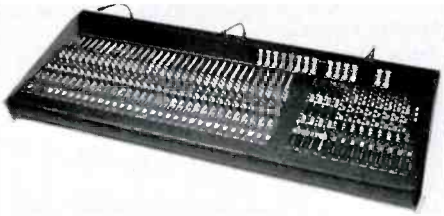
Price Range: $1,999.

INTERFACE ELECTRONICS

6710 Adler
Houston, TX 77081
Phone: (713) 660-0100

Models 32X8/32T8

Inputs: 16 to 48.
Bus Outs: Eight to 24.
Monitor Selection: Solos throughout.
Auxiliary Sends: Four.



EQ Section: Fixed or parametric.
Metering: Moving coil or LED VU.
Automation Capability: VCA option.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: 0.03% to +20 VU.
Signal-to-Noise Ratio: Better than -90 dB below zero VU typical.
Selected Standard Features: Transformer inputs; eight submixes with masters; any number of matrix outputs with VCAs for subgroupings and slider.
Price Range: $11,000 to $38,000

Models 200B

Inputs: Eight or 12.
Bus Outs: Two.
Monitor Selection: Solos throughout.
Auxiliary Sends: Two.
EQ Section: Three frequencies.
Metering: Three-inch standard VU.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: 0.03% to +20 VU.
Signal-to-Noise Ratio: Better than 90 dB below zero VU typical.
Selected Standard Features: Rechargeable battery power; available AC supply; 48v and 12V mike powering; many options.
Price Range: $3,900 to $5,600

Models 310B

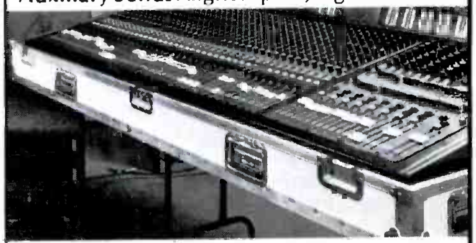
Inputs: 16 to 48.
Bus Outs: 10.
Monitor Selection: Matrix mixer.
Auxiliary Sends: Matrix mixer.
EQ Section: Four EQs, two parametric.
Metering: LED VU.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz, ±1 dB.
Distortion: 0.03% to +20 VU.
Signal-to-Noise Ratio: Better than -90 dB below zero VU typical.
Selected Standard Features: Stage monitoring mixer, expandable.
Price Range: $5,000 to $10,750

JIM GAMBLE ASSOCIATES

P.O. Box 7047
Tahoe City, CA95740
Phone: (916) 583-0138

Models SC32-16/HC40-24

Inputs: 32 and 40, respecively.
Bus Outs: 16 and 24, respectively.
Monitor Selection: N/A.
Auxiliary Sends: Eight inputs, eight returns.



EQ Section: Inputs: three fully parametric; outputs: four fully parametric.
Metering: 19 LEDs.
Automation Capability: N/A.
Frequency Response: 2 Hz to 150 kHz, -3 dB.
Distortion: 0.015% THD at 1 kHz, typical.
Signal-to-Noise Ratio: Better than -100 dB at 1 kHz.
Selected Standard Features: 30-band digital filter spectrum analyzer display on LED VUs; patchbay mike scrambling.
Price Range: SC32-16 (stage monitor): $60,000; HC40-24 (house console): $70,000

NEPTUNE ELECTRONICS, INC.

934 NE 25th Avenue
Portland, OR 97232
Phone: (503) 232-4445

Models 224XM and 164XM

Inputs: 24 and 16, respectively.

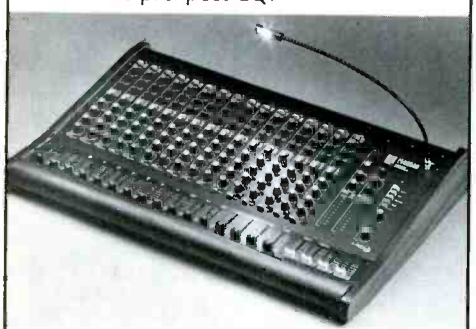
The Directory

Bus Outs: 10.
Monitor Selection: One, pre EQ.
Auxiliary Sends: One effects send, pre EQ; auxiliary send switchable pre/post EQ.
EQ Section: Three-band; sweep mid, shelving bass, switchable pre/post EQ.
Metering: Four-segment LED.
Automation Capability: N/A.
Frequency Response: 20 Hz to 20 kHz.
Distortion: Less than 0.2% at 20 Hz to 20 kHz.
Signal-to-Noise Ratio: –123 dB.
Selected Standard Features: Mike/line switching on each channel; external power supply; transformer balanced inputs; electronically balanced outputs; channels directly assignable to subgroups 1, 2, 3, 4 or direct to mains.

Price Range: 224XM: $5,195; 164XM: $3,695 (both with external PSU).

Models 1622/1222/8222

Inputs: 16/12/8.
Bus Outs: Six.
Monitor Selection: One monitor send; switchable pre/post-EQ.



Auxiliary Sends: One auxiliary send pre-post EQ and one effects post-EQ.
EQ Section: Three-band.
Metering: Two LED ladders.
Automation Capability: N/A.
Frequency Response: ±1 dB, 20 Hz to 20 kHz.
Distortion: Less than 0.2% at 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –90 dB, mic input to any output.
Selected Standard Features: Mike/line switching on each channel; differential balanced inputs; electronically-balanced outputs.
Price Range: 1662: $1,825; 1222: $1,525; 822: $1,025.

NEOTEK
1154 West Belmont
Chicago, IL 60657
Phone: (312) 929-6699

Series I

Inputs: Six to 48.
Bus Outs: Four or eight, plus optional stereo bus.
Monitor Selection: Six × two to 48 × eight × two.
Auxiliary Sends: Four.
EQ Section: Three-band with parametric midrange, or four-band parametric EQ.
Metering: 22 segment LED bargraph; optional mechanical VU.
Automation Capability: "Any system currently available."
Frequency Response: 20 Hz to 20 kHz, +1 dB/–0.5 dB.
Distortion: Less than 0.1%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –95 dB, 20 Hz to 20 kHz.
Selected Standard Features: "Built to individual order."
Price Range: From $9,500

Series II

Inputs: 16 to 56.
Bus Outs: Eight plus stereo/mono optional.
Monitor Selection: Six or 24 tracks in-line.
Auxiliary Sends: Four.
EQ Section: Four-band parametric.
Metering: Peak 22 segment LED bargraph and mechanical VU.
Automation Capability: "Any system currently available."
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: Less than 0.1%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –95 dB, 20 Hz to 20 kHz; typically 0.005%.
Selected Standard Features: Integral patch bay; two effects returns; two solo modes; built to order.
Price Range: From $19,000

Series III

Inputs: 28 to 56.
Bus Outs: 24 or 32, plus optional stereo bus.
Monitor Selection: 24 or 32 tracks in-line.
Auxiliary Sends: Six.
EQ Section: Four-band parametric.
Metering: Peak 22 segment LED bargraph and mechanical VU.
Automation Capability: "Any system currently available."
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: Less than 0.1%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –95 dB, 20 Hz to 20 kHz.
Selected Standard Features: Status logic; two solo modes, plus two in-place modes; two mute groups; integral patch bay; "built to order."
Price Range: From $40,000

Production Console

Inputs: 28 to 56.
Bus Outs: Eight to 48
Monitor Selection: Eight × four × two to 48 × eight × two.
Auxiliary Sends: Four.
EQ Section: Four-band parametric, or three-band parametric midrange.
Metering: Peak 22 segment LED bargraph and mechanical VU.
Automation Capability: Audio follow Video synchronizer interface.
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: THD less than 0.01%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –95 dB, 20 Hz to 20 kHz; typically 0.005%.
Selected Standard Features: Two echo returns; programmable start pulses; studio-control mute; overpress cue; P&G faders; "built to order."
Price Range: From $11,000

Theatre Systems Console

Inputs: Eight to 56.
Bus Outs: N/A.
Monitor Selection: Eight × four; 56 × 48.
Auxiliary Sends: Four.
EQ Section: Three-band parametric midrange, sweep high-pass filter.
Metering: Peak 22 segment LED bargraph and mechanical VU.
Automation Capability: "Any automation currently available."
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: THD less than 0.01%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –90 dB, 20 Hz to 20 kHz.
Selected Standard Features: 14 presets; 11 user-definable signal flow options; "built to order."
Price Range: From $11,000

Elite

Inputs: 32 to 56.
Bus Outs: 26 plus stereo and optional mono.
Monitor Selection: 26 tracks in-line.
Auxiliary Sends: Six.
EQ Section: Four-band parametric plus tuneable high pass.
Metering: Peak 22 segment LED bargraph and mechanical VU.
Automation Capability: "Any automation currently available."
Frequency Response: 20 Hz to 20 kHz, ±0.1 dB.
Distortion: THD less than 0.05%, 20 Hz to 20 kHz.
Signal-to-Noise Ratio: Better than –95 dB, 20 Hz to 20 kHz.
Selected Standard Features: Integral patchbay; three solo modes; three mute groups; three in-place solo modes.
Price Range: From $36,800

RUPERT NEVE, INC.
Bershire Industrial Park
Bethel, CT 06801
Phone: (203) 744-6230

542 Series

Inputs: Six to 16.
Bus Outs: Two or four.
Monitor Selection: Stereo/mono.
Auxiliary Sends: Two.
EQ Section: Three-band with HP filter.
Metering: Two or six; VU/PPM.
Automation Capability: VCA option for video editing.
Frequency Response: 15 Hz to 20 Hz,

+0.5/–1 dB.
Distortion: Less than 0.05% at 20 dBu output.
Signal-to-Noise Ratio: Better than –125 dBu EIN.
Selected Standard Features: Balanced inputs and outputs; transformer isolated; XLR connectors.
Price Range: $9,600 to $26,300

5104 Series

Inputs: 16 to 60.
Bus Outs: Four.
Monitor Selection: Stereo/mono.
Auxiliary Sends: Four.
EQ Section: Two- or four-band, plus filter.
Metering: VU or PPM.
Automation Capability: NECAM 96 Servo Fader disk-based system.
Frequency Response: 20 Hz to 20 Hz, ±0.5 dB.
Distortion: Less than 0.03% at 20 dBu output.
Signal-to-Noise Ratio: Better than –125 dBu EIN.
Selected Standard Features: Stereo television production and post production; modular design; selection of input-module types.
Price Range: From $40,000

5106 Series

Inputs: 24 to 60.
Bus Outs: Eight.
Monitor Selection: Stereo/mono.
Auxiliary Sends: Eight.
EQ Section: Four-band, plus filters.
Metering: VU or PPM.
Automation Capability: NECAM 96.
Frequency Response: 20 Hz to 20 Hz, ±0.5 dB.
Distortion: Less than 0.03% at 20 dBu output.
Signal-to-Noise Ratio: Better than –125 dBu EIN.
Selected Standard Features: Stereo television production and post production; compressor/limiter/noise gate on each input.
Price Range: From $64,000

Model 8128/8128-TV

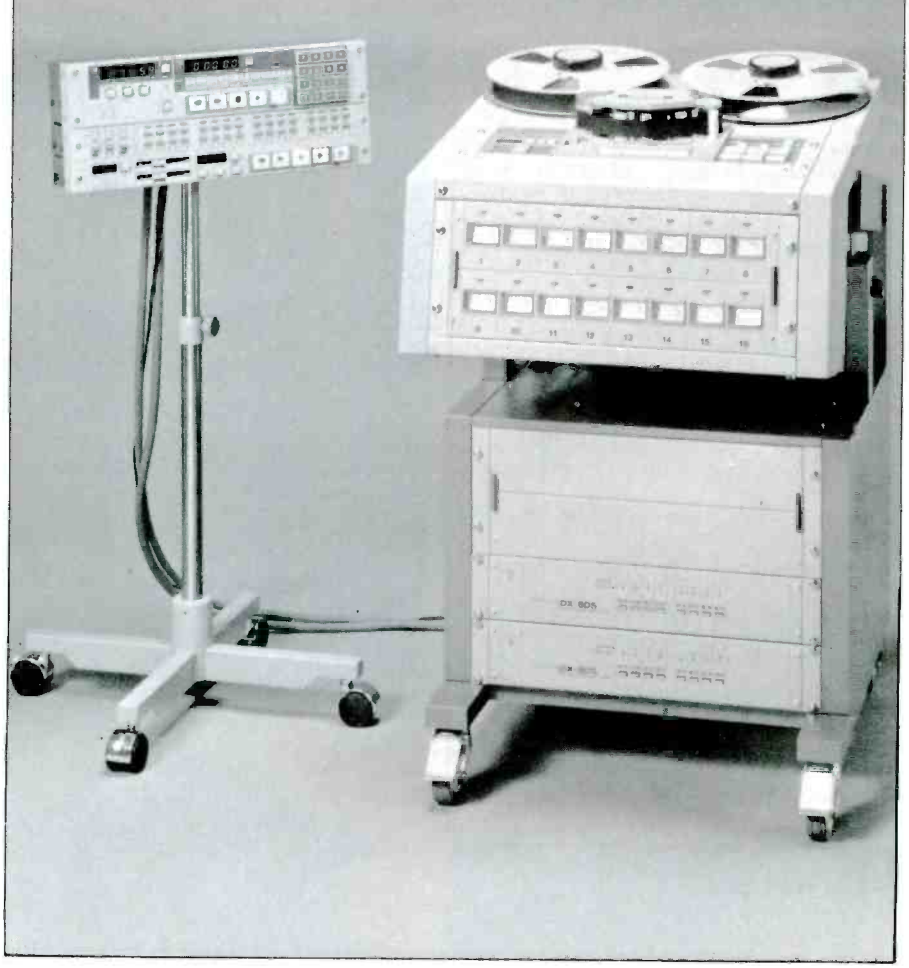
Inputs: 32 to 56.
Bus Outs: 24 to 48; 8128-TV has four stereo pairs.

Monitor Selection: In-line and eight-track.
Auxiliary Sends: Four mono, one stereo.
EQ Section: Four-band FSE, plus filters.
Metering: VU and PPM 200 segment bargraph.
Automation Capability: NECAM 96.
Frequency Response: 20 Hz to 20 Hz, ±0.5 dB.
Distortion: Less than 0.06% at 20 dBu output.
Signal-to-Noise Ratio: Better than –124 dBu EIN.
Selected Standard Features: 8128 is a studio master recording console with central track assignment and memory recall; 8128-TV is a Stereo TV and film post post-production sweetening console with four dedicated stereo busses for music/effects-dialog/laugh track.
Price Range: 8128: $115,000; 8128-TV $122,000

Part Two of this Product Listing, to be published in the next issue, will include details of console products from Panasonic/RAMSA, Pulsar, Rane, Roland, Ross Systems, Solid State Logic, Sony, Soundtracs, Soundcraft, Sound Workshop, Spectra Sonics, Studer, Studiomaster, TOA, Trident, Wheatstone, Yamaha, and a host of others.

TASCAM MS-16 ONE-INCH 16-TRACK

Reviewed by Peter Butt



The Tascam MS-16 is a low-cost, 16-track machine using a one-inch-tape format at a single speed of 15 ips, available in either a table-top or rack-mount unit, or may be mounted in an optional mobile console. An optional remote control and autolocator are also available. The dbx noise reduction systems supplied and tested with this machine also are optional, and may be bypassed with multipin shorting plugs located at the rear of the amplifier case.

The MS-16 is not typical of what I have been conditioned to expect of the "narrow-gauge" class of audio systems that have come along since the early Seventies. The outer trim is an apparently heavy coating of semi-gloss beige and brown enamel. The deck plate is made of approximately 3/16-inch aluminum plate, as opposed to the bent steel sheet commonly encountered. The high-level XL-type audio connectors accept a true +4 dBv input with fair common-mode rejection. The XL-type outputs feature a 20-ohm source impedance in a quasi-balanced, push-pull, DC-coupled dual-active driver. The term "quasi-balanced" is used here, since any shorting connection between either of the output signal pairs to ground will result in significant ground currents, impairing the slewing performance of the output circuits. It is possible that the shorted driver may experience strain to the point of premature failure, should the short persist for a very long time, signal or no. The suggested method of driving an unbalanced (single-ended) load is to simply ignore the LO side of the MS-16 high-level output pair, and connect between the HI and ground.

This procedure would seem to be somewhat dissatisfying. In doing this, there is the immediate loss of 6 dB in signal level and headroom. Many uninformed users could be tempted to connect or patch any output to any following input without regard for technical details. Clearly, disappointment or even sadness could result.

The deck features an electronic tape-time (footage) counter having a Return-to-Zero and a Search-to-Cue capability. The capstan-drive system relies on the *de-facto* standard 9,600 Hz, frequency-modulated TTL level signal for external speed control. All deck functions are accessible via rear-panel multipin connectors. A front-panel tape-lifter defeat switch, stop mute, and lifter defeat mute are providedon the front panel. Track #16 can be excluded from the mute functions as it is intended to be used as a timecode track.

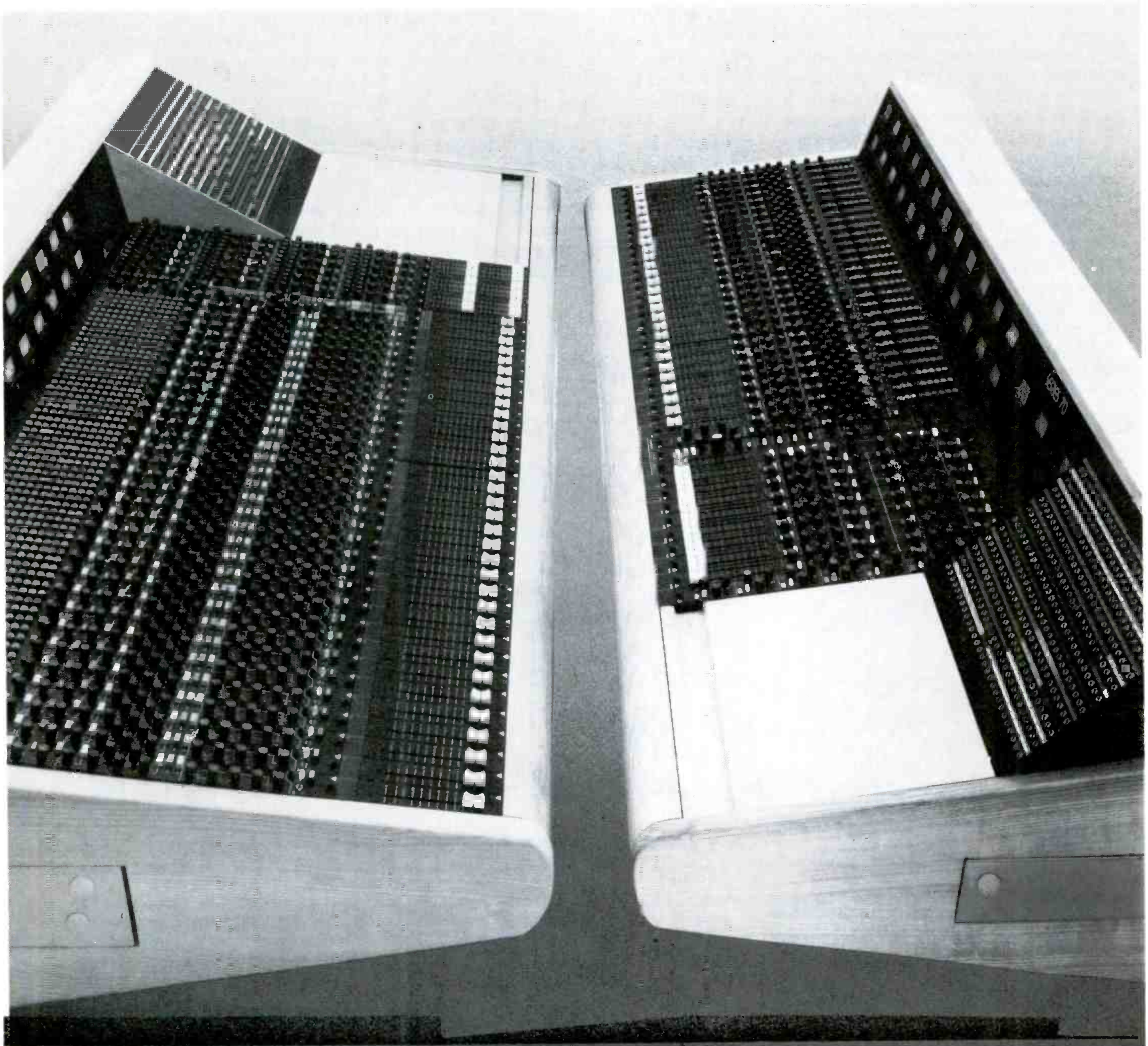
External synchronizer control is interfaced through an ELCO 8016-type, 38-pin connector with all the usual in/out functions including a tape-counter tach pulse signal.

The machine behaves quietly in all modes. Deck cooling is enhanced by a DC fan that is substantially silent. A spooling mode is accessed by a double press of either fast forward or rewind buttons, or by logic command from an external synchronizer, if used. Individual channel-status controls are located on the lower part of the deck front panel, below the head assembly (wired to multipin connectors!) and stock one-inch editing block. The audio-channel status may be locally or remotely determined, either individually or as ganged functions. The audio channels may be switched to Input Monitor automatically during periods of shuttle or stop mode operation.

Tape tensions are controlled by DC servo systems that operate as constant-tension systems. Reel-motor braking is of the solenoid operated band-type. The machine is normally supplied with plastic NAB locking hubs that may be replaced by heavy-duty, all-metal hubs if necessary.

AC power is supplied through a fairly heavy two-wire, captive cord having a non-polarized, two-prong

.....tried, trusted, TRIDENT



The Series 80B is a 32 input. 24 group/monitor console, and the Series 70 is a 28 input, 16 group 24 monitor console. Both desks are fitted with a fully professional patchbay, 4 echo returns with EQ, and have the facility to use the monitor section as further inputs on remix.

Your limit isn't 32 inputs. Trident's engineers will design you an individual 56 input console at a price which will impress your bank manager.

So don't mince words – say Trident.

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(503) 239-0352

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(404) 436-3024

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(602) 267-0600

Audio Techniques
New York City
(212) 586-5989

TRIDENT

TRIDENT U.S.A. INC.

280 Mill Street Ext, Lancaster, Mass 01523 U.S.A.
Tel: (617) 365-2130/(617) 368-0508 Telex: 951546

See Us at AES in booth number 341-343

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plug. The U-type, three-wire grounding plug would be more desirable for this, or indeed, any machine.

The audio electronics are contained in a separate case. External audio interface connectors are mounted on the rear panel in two sets: 16 pairs of XL-type connectors on one side, and 16 pairs of RCA-type phono connectors on the other. Head assembly connections are made to the audio amplifier case through captive cables from the deck that connect to the amplifier case via three multipin connectors. Audio channel status commands are carried from the deck assembly to the audio amplifiers by a multipin connector/cable assembly.

The audio-level metering is small but readable and illuminated. Meter accuracy is sufficient to be useful for most purposes, as is shown in Table 1.

Access to the record/play adjustments is made by dropping the front meter panel after removing four socket head capscrews. The meter panel is hinged in such a way that it can be viewed from either above or below while adjustments to the audio amplifiers are being made.

Insertion of the dbx noise-reduction systems is achieved by removal of two multipin shorting plugs at the rear of the amplifier case, and connecting the dbx case-captive cables to the exposed ports. Each dbx case accommodates eight channels of record/play, dbx Curve-1 noise reduction with bypass capability on an individual channel basis.

System Performance

MS-16 high-level input monitor response is shown in Figure 1; the curves shown are typical. A 10-kohm IHF load was used for all the measurements shown, since it is considered extraordinary service to encounter a true 600-ohm loading condition in this age of bridging input devices. Although the MS-16 will drive a 600-ohm resistor from its high-level outputs ports, this should never be required of it. The curves are shown for the case of dbx-in and dbx-out. The dbx magnitude curves are separated from their companions by a 1 dB offset.

The response for the single-ended phono ports is shown in Figure 2. The curves are very similar to those of the high-level curves of Figure 1, except that the channel response droops a little less at about 5 Hz than for the high-level ports. Insertion of dbx noise reduction into the channels is evident by the slight shelve-up below 200 Hz, and the more severe band limiting and phase shift as compared

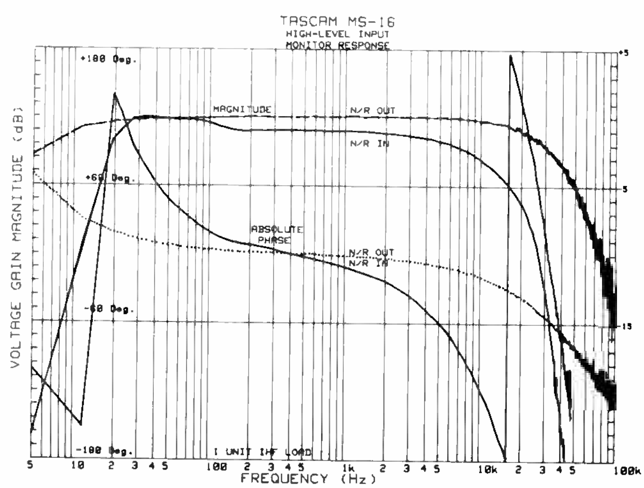


Figure 1: High-level input monitor response.

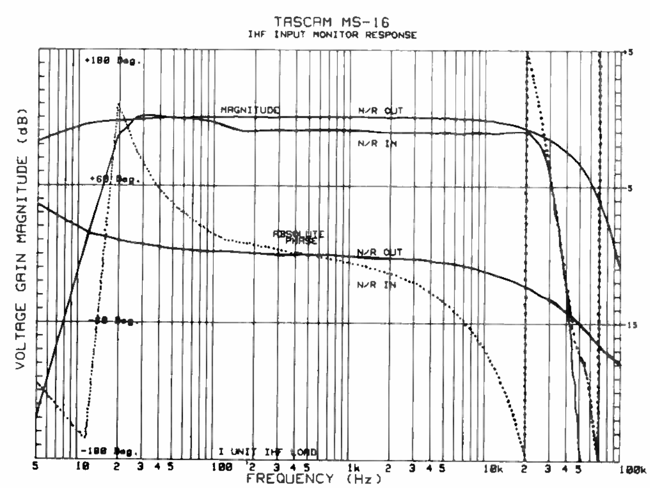


Figure 2: IHF input monitor response.

Figure 3: High-level input monitor Absolute Group Delay response.

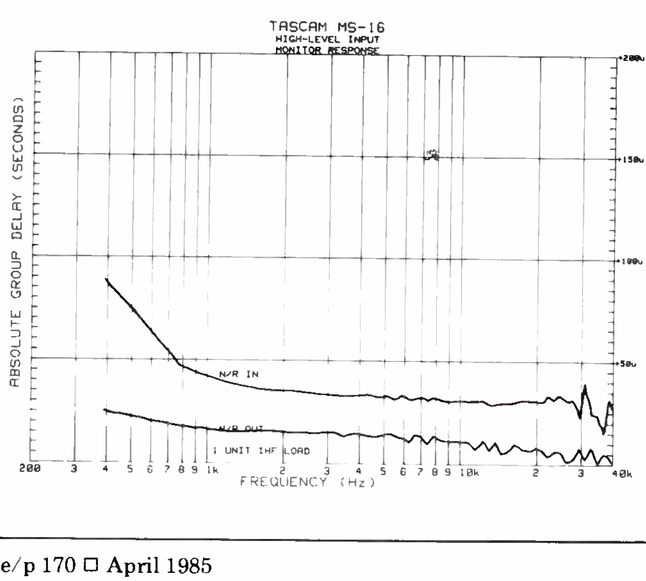
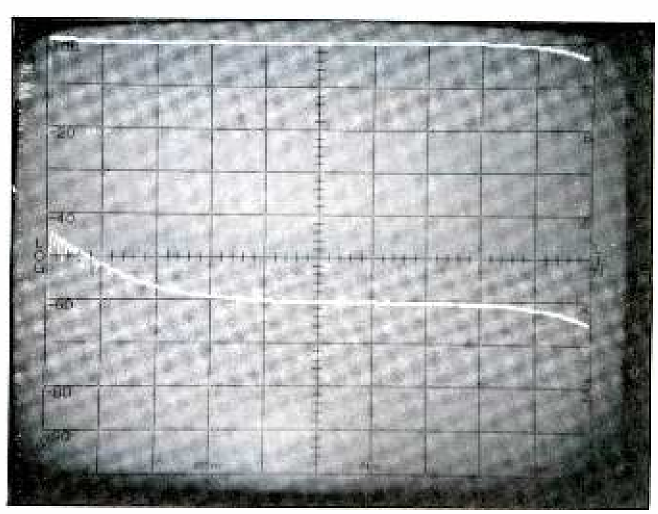


Figure 4: High-level input port common-mode response. Top trace is normal-mode response, the lower trace is common-mode response. Frequency scale is logarithmic, 10 dB per division, vertical; 30-Hz resolution bandwidth. Top graticule reference is 0 dBv.



A BIG PART OF OUR MICROPHONE TEAM COMES IN EACH BOX

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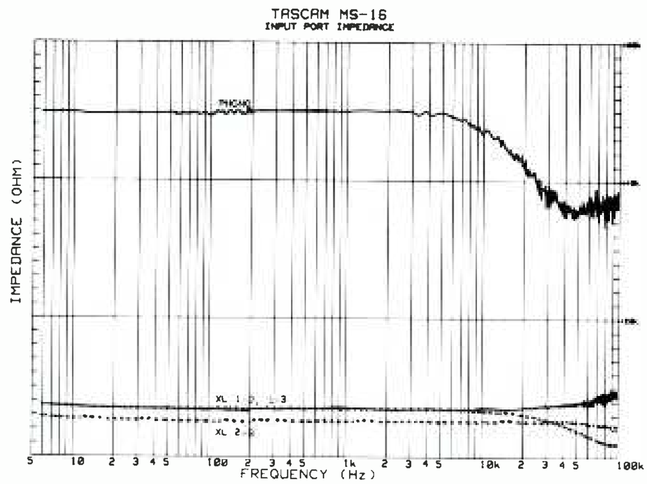


Figure 5: Input port impedances.

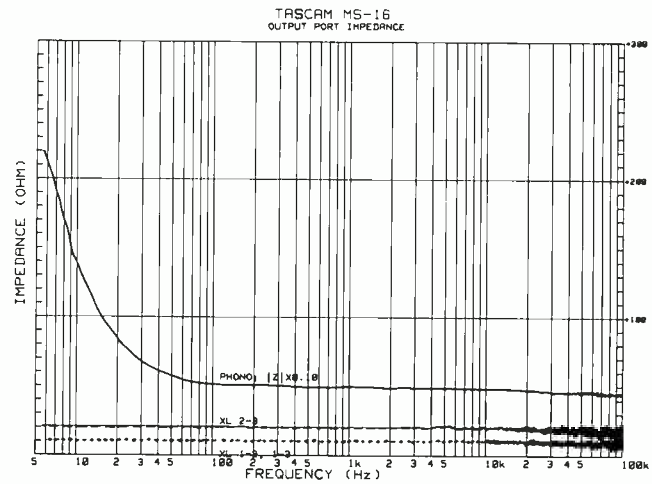


Figure 6: Output port impedances.

TASCAM MS-16 REVIEW

with the N/R OUT traces. It is apparent that the audio path is encoded and decoded when noise reduction is activated without regard to the audio channel status. High-level port phase shift is slightly greater than for the phono ports, because the high-level electronics function as cascaded buffers, transforming the high-level signal to the IHF level at the inputs, and IHF level to +4 dBv at the outputs.

Group delay curves for the high-level input monitor data are shown in Figure 3; they are sufficiently boring to be considered quite good. Their flatness is excellent even considering their slight slopes, showing decreasing delay with increasing frequency. The input monitor channels display a nearly linear phase response even if noise reduction is selected.

The log-frequency sweep of Figure 4 shows the common-mode rejection typical of the high-level differential input ports. The top line is the normal mode response reference, while the lower trace is the common-mode response, showing below 50 dB for frequencies above about 40 Hz, and -60 dB or better above about 500 Hz.

Typical input-port impedances versus frequency are shown in Figure 5. The input ports generally behave as

. . . continued overleaf —

Table1: Typical Meter Response Input level for Constant VU Indication versus Frequency

Frequency (Hz)	Relative Input Level (dB)
10	+1.30
20	+0.45
30	+0.19
40	+0.15
50	+0.02
60	0.0
70	0.0
80	0.0
90	0.0
100	0.0
500	0.0
1k	0.0
2k	-0.05
4k	-0.05
8k	0.0
10k	+0.05
12k	+0.15
14k	+0.20
16k	+0.25
18k	+0.40
20k	+0.50
25k	+0.95
30k	+1.40
35k	+1.93

Input Monitor Mode; dbx Noise Reduction Switched Out.

SUMMARY OF TASCAM MS-16 SPECIFICATIONS

Description	Quoted	Observed
Track Config:	16-track, 1-inch	Yes
Tape Speeds:	15 ips; 381 mm/s	Yes
Reel Capacity:	Up to 10.5 inch (305 mm); NAB hub.	Yes
Drive:	Servoed DC reel and capstan motors; Pinch-roller drive, 9,600 Hz speed reference; external reference selectable.	Yes
Fast Wind Time:	120 seconds for 10.5-inch reel; 2,400 feet.	115 seconds for 10.5-inch reel; 2,475 feet.
Spooling Wind Time:	370 seconds for 10.5-inch reel; 2,400 feet.	364 seconds for 10.5-inch reel; 2,475 feet.
Speed Accuracy:	0.2% deviation.	±0.051% head to tail; 10.5-inch reel, 2,475 feet.
Wow and Flutter:	15 ips: less than ±0.08%; peak DIN WTD; ±0.07% peak DIN UNWTD; 0.04% NAB WTD; 0.07% NAB UNWTD.	±0.035% DIN WTD; ±0.130% DIN UNWTD (mean of 5-30s samples).
Head Config.:	Three-head: erase, record/sync, reproduce.	Yes
Tape Cue:	Manual and automatic (RTZ and STC).	Yes
Pitch Control Coarse:	±15%; continuously variable	+17.44%, -17.035%; all speeds.
Pitch Control Fine:	±0.7% continuously variable	+0.945%; -0.858%
Dimensions (W×H×D):	Transport unit: 482×459×310 mm; 19×18×12.2 inches; Amplifier unit: 482×193×321 mm; 19×17.3×12.3 inches;	
Weight:	Transport unit: 38 kg; 83.75 pounds; Amplifier unit: 16.5 kg; 36.38 pounds.	

ELECTRICAL

Description	Quoted	Observed
Line Input:		
Balanced:	Balanced.	Imbalanced differential single op-amp.
Input Impedance:	10 kohms.	5.18 kohms at 1 kHz; 6.81 kohms at 1 kHz; each side of line to ground.
Nominal Input Level:	+4 dBV (1.23 Vrms).	+4 dBv (1.23 Vrms).
Maximum Input Level	+28 dBv (19.5 Vrms), balanced.	+30.4 dBv (25.5 Vrms).
Unbalanced:	Unbalanced.	Single-end, case grounded.
Input Impedance:	50 kohms.	52 kohms.
Nominal Input Level:	-10 dBV (0.316 Vrms).	-10 dBV (0.316 Vrms).
Maximum Input Level:	+18 dBV (8.0 Vrms).	+16.7 dBV (6.8 Vrms).
Line Output:		
Balanced:	Balanced.	Push-pull, low-Z power driver.
Output Impedance:	20 ohms.	20.3 ohms, at 1 kHz, floating load; 10.7 ohms at 1 kHz each side of line to ground.
Nominal Output Level:	+4 dBv (1.23 Vrms).	+4 dBv (1.23 Vrms).
Maximum Output Level:	+28 dBv (19.5 Vrms).	+30.0 dBv (24.5 Vrms).
Unbalanced:	Unbalanced.	Single-end, case grounded.
Output Impedance:	500 ohms.	500 ohms at 1 kHz.
Nominal Output Level:	-10 dBV (0.316 Vrms).	-10 dBV (0.316 Vrms).
Maximum Output Level:	+18 dBV (8.0 Vrms).	+16.7 dBV (6.8 Vrms).
Bias Frequency:	145 kHz.	143.416 kHz.
Equalization:	IEC, infinity to 35 microsec.	IEC, infinity to 35 microsec.
Calibration Level:	250 nW/m.	250 nW/m.
Frequency Response:	15 ips.	
Record/Play:	40 Hz to 22 kHz, ±3 dB.	27 Hz to 28 kHz, ±3dB.
Frequency Response:	15 ips.	
Record/Sync:	40 Hz to 22 kHz, ±3 dB.	30 Hz to 30 kHz, ±3 dB.
Third Harmonic Dist.:	0.6% at 250 nW/m, 1 kHz.	0.22% at 250 nW/m, 1 kHz; 4.37% at +13 dB re: 250 nW/m, 1 kHz; 3M 226.
Signal-to-Noise:	69 dBA (NAB), dbx out; 62 dB UNWTD, 0 to 100 kHz; 107 dBA (NAB), dbx in; 100 dB UNWTD, dbx in.	Re: 3% 3HD, 1 kHz UNWTD; 20 Hz to 20 kHz BPF; Repro: 66.8 dB, dbx out; Re: 3% 3HD, 1 kHz UNWTD. 20 Hz to 20 kHz BFP; Repro: 96.4 dB, dBx in; Sync: 94.6 dB, dBx in.
Adjacent channel Crosstalk:	Greater than 55 dB down at 1 kHz; 0 VU.	At 0 VU indicated, -64, -66 dB adjacent tracks; Record/Repro mode, 1 kHz; dbx out.
Erasure:	Better than 70 dB down at 1 kHz; +10 VU ref.	Greater than 80 dB down at 1 kHz; +10 VU ref.
Power:		
USA/Canada:	120 VAC, 60 Hz	Yes. non-polarized, two-wire plug; Captive power cord.

. . . continued overleaf —

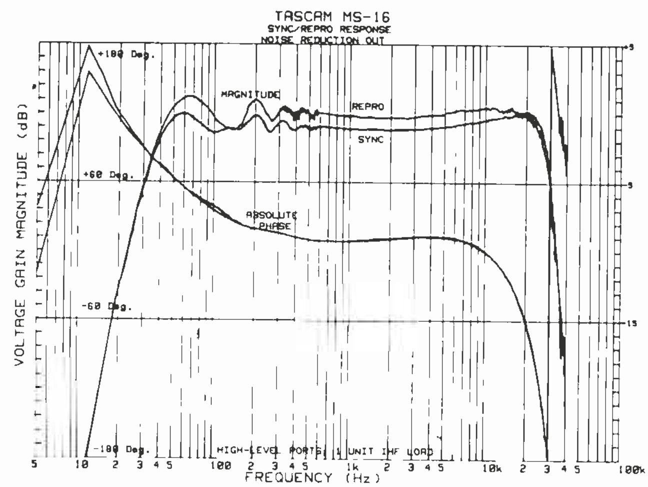


Figure 7: Sync/Repro response; Noise reduction out.

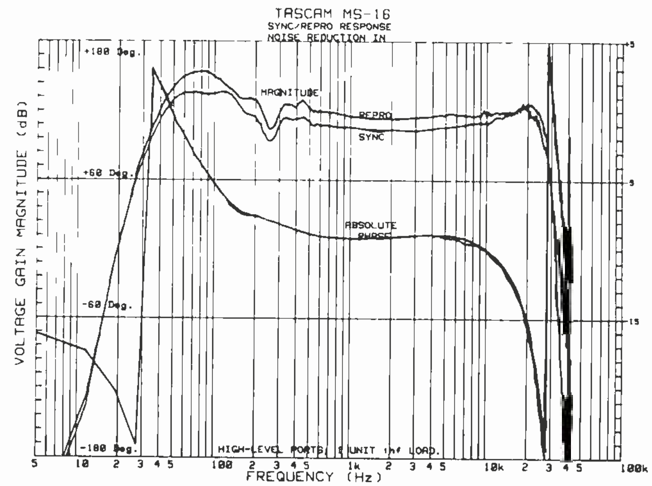


Figure 9: Sync/Repro response; Noise reduction in.

TASCAM MS-16 REVIEW

resistances over the significant portions of their audio range. The ports should not be sensitive to source impedances below about 500 ohms. Lower driving impedances would be still more desirable.

Output-port impedance behavior is shown in Figure 6. The high-level outputs show substantially DC coupling and a constant impedance extending beyond 100 kHz. The phono output impedance is AC-coupled, with a distinct rise below 100 Hz. Note that the PHONO curve is plotted multiplied by 0.1 so that it can be clearly shown on the same graph with the high-level port data.

The record/play transfer function of the MS-16 is shown in Figure 7. Record/repro and record/sync magnitude and phase curves without noise reduction are shown overlaid, the magnitudes being offset by 1 dB. Magnitude responses of both modes are very similar. Contour effect

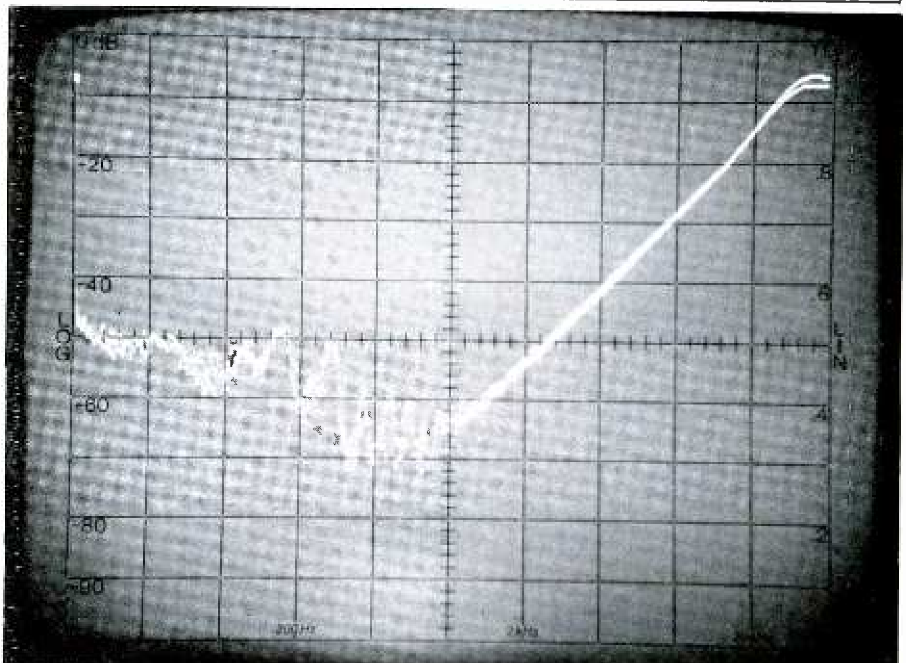


Figure 8: MS-16 flux-loop response. Top trace is sync-head response, the lower trace is reproduce-head response. Logarithmic frequency sweep, 10 dB per division, vertical; 30-Hz resolution bandwidth.

SUMMARY OF TASCAM MS-16 SPECIFICATIONS

Description	Quoted	Observed
Europe: UK/AUX	220/240 VAC, 50 Hz.	
General Export:	100/120/220/240 VAC; 50/60 Hz.	
Power Consumption:	80W.	
Additional		
Interface:		
Audio:	Three-pin XLR-type and RCA-type phono connectors.	Yes.
Remote control:	Multipin connector.	Yes.
Function remote:	Multipin connector.	Yes.
Accessory:	Multipin connector.	Yes.
dbx unit:	Multipin connector.	Yes. Two connectors for 16 channels; two dbx units.
Audio signal polarity:		Positive-going input at XL pin #3 yields positive-going signal at RCA-type phono ports and at output XL pin #3. All ports erect; XL pin #3.
Magnetic polarity:		Negative. Record/sync/play erect.
Hardware:		ISO metric.

Price: Suggested retail price: $8,995;
DX-8 dbx NR unit (eight-channel): $849;
Manufacturer: TASCAM, 7733 Telegraph Road, Montebello, CA 90640. (213) 726-0303.

extends significantly up to about 500 Hz, being somewhat less severe for sync reproduction. There is a slight hint of sync-head resonance, but not enough difference to be significant as far as audibility is concerned.

A log sweep of repro- and sync-head loop response is shown in Figure 8. The traces are overlaid, with the slightly under-damped sync-head peak extending noticeably above the repro-head curve. Both heads self-resonate at about 35 kHz. Each repro pre-amp has a padding capacitor that may be trimmed by substitution to achieve uniform head resonances upon head replacement.

The record/repro and record/sync data taken with the dbx noise-reduction system engaged is shown in Figure 9. Comparisons with Figures 7, 1 and 2 show that low-end shelving of the dbx system is uniformly evident in all aspects of MS-16 performance. The head-contour bumps are emphasized by the noise-reduction, as are all deviations from flat response. The other impact of noise-reduction insertion is a phase rotation increase with frequency, due primarily to the band limiting that is essential to reliable operation of the dbx companding system.

Record/play, record/sync group delay response for the cases of dbx-in and -out of the system are shown in Figure 10. The consequences of the dbx record/reproduce complementary boost/cut can be seen in the increasing group delay with increasing frequency in all modes. There is no attempt to correct audio signal phase in the MS-16. The careful damping of head resonances has been effective in minimizing the serious phase distortion that often results from inadequate attention to this detail of analog recorder performance. Group delay is only slightly degraded with noise reduction engaged.

Interchannel crosstalk is always an annoying feature of analog system performance. Figure 11 shows the crosstalk transfer from one track to each of its adjoining neighbors. Track #2 record/reproduce magnitude at nominal zero VU indicated record level is shown for reference as the top trace. Scale factors for the figure are 10 dB per division, vertical, and 5 kHz per division, horizontal. Crosstalk of track #2 to tracks #1 and #3 are shown centrally, and system residual noise and stray leakage as the bottom trace. The overall performance is quite acceptable across the band, peaking at -45 dB or less at about 25 kHz in each case. The audibility of this kind of crosstalk characteristic will depend on the nature of the program involved. Such a degree of channel isolation is unusually good and quite unexpected

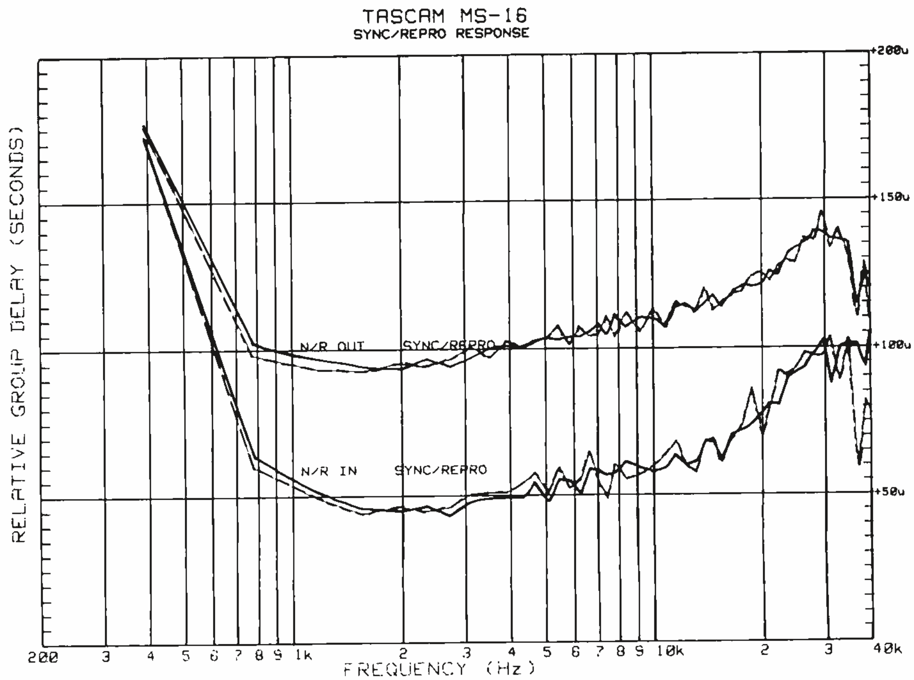


Figure 10: Sync/Record Relative Group Delay response.

in a tape-track geometry as crowded as this one is.

The 2:1 companding ratio of dbx noise reduction will tend to emphasize the magnitude of level variations arising within the signal transmission channel. The increased degree of rolloff and peaking observable in the noise-reduction data of Figure 9 are symptomatic of this effect. As a check of the impact of this kind of level exaggeration, a 10-kHz tone was recorded and replayed with dbx noise reduction alternately switched in and

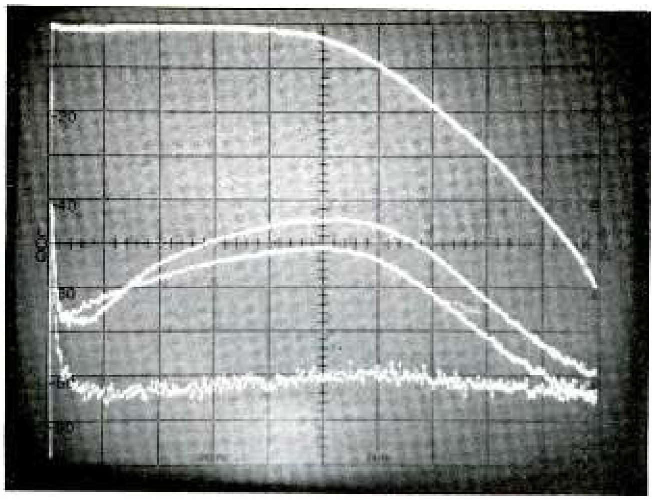


Figure 11: Crosstalk between adjacent tracks; record/reproduce mode. Top trace: track #2 record/reproduce response; center traces: track #1 and #3 reproduce signal leakage; bottom trace: system residual noise/leakage, stop mode, repro muted. Scale factors: 10 dB per division vertical; 5 kHz per division linear horizontal. Top graticule reference level: -10 dBV, nominal zero-VU record level. 100 Hz resolution bandwidth, 5s seconds per division sweep speed. Analyzer LO offset: -115 Hz.

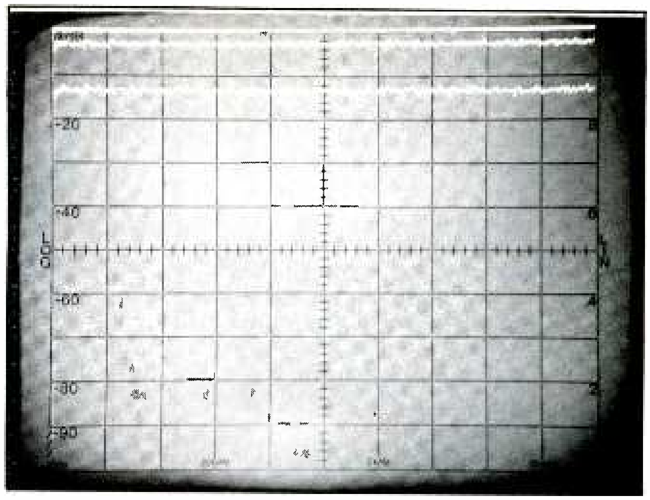


Figure 12: 10-kHz record/playback level versus time; 10-kHz modulation frequency. Top track: line input; center trace: reproduce, dbx-out; lower trace: reproduce, dbx-in. Vertical scale: 2 dB per division; horizontal scale: 5 seconds per division. 100 Hz resolution bandwidth.

out for successive passes; results of this test are shown in Figure 12. The top trace is the level of the generator signal as observed through the MS-16 input monitor mode; scale factors are 2 dB per division, vertical, and 5 seconds per division, horizontal. The middle trace is the MS-16 playback with dbx switched in. The worst-case level deviations are about 0.4 dB peak-to-peak or less, a figure that would likely be acceptable for most users.

MS-16 scrape-flutter performance is shown in Figure 13, and comprises the reproduction spectrum of a 10-kHz tone over a 200-Hz range centered on the signal frequency. Low-frequency flutter sidebands are the most prominent in the immediate region of the fundamental signal peak. Flutter components more than 60 Hz away from the signal are attenuated more than 40 dB below the fundamental peak, indicating that the scrape-flutter components lie more than 50 dB

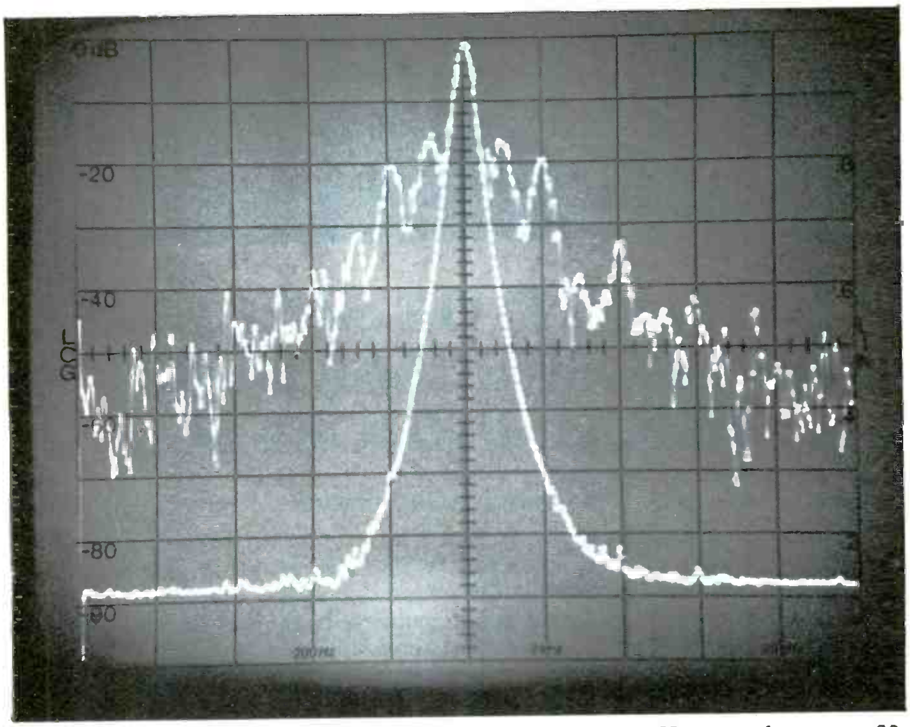


Figure 13: 10-kHz modulation record/reproduce spectrum. 10-kHz center frequency; 20 Hz per division horizontal; 10 dB per division vertical; 3-Hz resolution bandwidth. Sweep speed: 5 seconds per division. Line input sweep overlaid.

below the signal. The shape of the spectrum analyzer 3-Hz filter is shown as a trace of the modulating signal observed through the MS-16 line input monitor, overlaid on the reproduced signal spectrum.

Summary

The Tascam MS-16 is a surprisingly good machine by any measure. It would seem that the "semi-pro" class of audio equipment has been derived largely by the addition of larger meters and a few more lights to what are substantially consumer-level devices. Serviceability and quality assurance — features that I consider to be the true delineators of the differences between expensive toys and useful tools for those of us whose livelihoods and reputations are on the proverbial line — have not been the hallmark of semi-pro gear. It is a pleasure therefore to find a comparatively inexpensive piece of equipment that doesn't make the routine checks and adjustments necessary to the achievement of a high degree of quality and reproducibility of audio program product a task rivaling a frontal lobotomy. Head assemblies can be interchanged with only a metric hex driver. Biasing and equalizing are performed with full availability of metered-level information. Speed offsets are easily read to 0.1% resolution and minimal Search-to-Cue/Zero functions make this machine an outstanding performer in the audio signal respects alone.

Add to this the accessibility of capstan-drive signals and external-control functions without evisceration of the control electronics, and we have a fine example for other competitors in this market to emulate. ■■■

IN-USE OPERATIONAL ASSESSMENT

EVENTIDE SP2016 DIGITAL SPECIAL EFFECTS SIGNAL PROCESSOR

Reviewed by Bob Hodas

Eventide has finally come out of the closet with its Effects Processor. You have probably seen the SP2016 displayed at various exhibitions during past couple of years, and I have even seen one in a studio. But the unit reviewed here is a new, upgraded version with improved bandwidth and additional software. The SP2016 is aptly named, since it is more than just a digital reverb: factory-loaded programs include plates, several mono and stereo rooms, delay lines and multi-tap lines, loop edit, flanging, chorusing, comb filtering, and more.

What I find most exciting about the SP2016 is something that separates it from every digital effects device on the market. The unit can be interfaced with an IBM or Hewlett-Packard computer to enable the user to modify or design his own software programs. The creative side of computers if finally opening up for the recording engineer, and he or she will no longer be at the mercy of what a manufacturer thinks is right for the market. It's not that the equipment manufacturers provide us with *bad* programs; only that, with the technology now available, more end-users will be able to contribute to state-of-the-art sound enhancements. But more on this personal observation later. First, let's take a look at the SP2016's operation.

Back-panel connections are more numerous and varied than other processors. There is the always necessary AC connector, with a voltage selector and an easily accessible fuse adjacent. Input and output connections are standard three-pin XLRs. There are two inputs with 10 kohm impedence, line level -10 to +24 dBm, and two outputs, 150 ohms capable of maximum output of +18 dBm. In and out connectors form part of a modular-design analog module that is designed as a plug-in subchassis, so that in the future the unit can accept an all-digital I/O module. The processor contains a bypass relay to route the signal directly through when the unit's power is off. Since many times it is all too easy to blame gremlins on the outboard equipment, this bypass mode will speed your problem tracing when experiencing signal loss or noise.

Two methods of remote control can be accessed on the rear panel: a DIN jack for the dedicated mini-remote control; and an IEEE-488 computer interface for normal remote functions, plus the control of up to 15 units (including other Eventide products), automated control for mixing, and programming your own effects.

Lastly, there is a control-voltage trigger input jack for external triggering of programs such as Loop Edit. The manufacturer has left room for growth in the unit's functions by including a special-purpose interface slot which, at this time, remains unoccupied.

Front-Panel Controls

Front-panel layout is neat, uncluttered and well labelled. The *Input Level* block contains two sliders for left and right input level control. (As with most digital equipment, it is best to have these set to the "off" position during power up). Between the sliders are two bargraph level displays; it is recommended that the input level be set to a point at which the limit indicator lights on occasional peaks. Such a setting will provide the best signal-to-noise ratio during operation. (As with all digital processing equipment, you don't want to light the Overflow indicator, or your ears are in for some harsh, unnatural distortion.) Below the bargraphs is a Monitor button that changes the level indicators from input to output. A momentary-action control that lasts for nine seconds, this control is handy for balancing input and output levels; the word "Output" lights just above the button to indicate its status.

The Output block contains four sliders: two assigned as left and right output level controls; and two for left and right Dry/Effect mix at the output. All front-panel sliders operate under digital control, and are completely programmable. As is the case with many digital faders, however, a slight "zippper" noise is sometimes audible when moving the control. The effect does not provide any problems to this reviewer, since once input and output levels have been established, all send and return levels are normally controlled from the console. The same applies with the mix sliders — the unit will normally be operated with Effect mix control fully open, using console returns to control how much of the effect goes to tape.

The Status block displays Input Mode or either Mono or Stereo, depending on the program running, and bandwidth of either 8 or 16 kHz (also program dependent, although most programs offer a full 16 kHz bandwidth). When the remote control is enabled, the word "Remote" is displayed, disabling the controls in the Processor Control block. If a red "Low Battery" lights up, you are being warned that you must replace the battery backup, or risk loosing all of your presets.

The *Processor Control* section comprises the heart of the SP2016. All information concerning programs and operation parameters is displayed in a window capable of holding 16 alphanumeric characters. (The display window size is roughly four by 0.25 inches high.) Upon power up, the window displays the words "Self-Test." In this mode, the unit tests all front-panel lights, and runs a minor diagnostic on the internal circuitry. (We will explore the self-test capability more extensively later.) After the self-test, a quick message indicates the number of programs and presets available, and than displays the currently-loaded program name.

Programmmable Features

A loaded program is indicated by a solid red light above the *Program* key. To search through the programs available, you simply press this key, and the appropriate names appear in the display window. While looking for a new program, the red light flashes indicating that what you see in the display window is not what you are hearing. Upon finding the new program you want to use, hit the *Execute* Key, the red light stops flashing and your new effects program is loaded from EPROM into the user memory workspace. The program search may be stepped through slowly by tapping the Program key or, more quickly, by

. . . continued overleaf —

holding the key down continuously. The program names scroll quickly (the full 84-program complement scrolls in 35 seconds) but are easily read. If you pass the desired program you can scroll backwards to find it again. The *Adjust/Select* slider can also be used to scroll program names, but I found it either too fast or too slow in operation.

Once an effects program is selected, hitting the *Parameter* key will scroll you through the various control parameters that are available for tailoring the program to your specific needs. If you have a question about a specific parameter, you need only hit the *Define* key and the display window gives you a description of that parameter. It's a great feature that saves time for those not so familiar with this unit, and who would otherwise have to page through the manual. By the way, the Define key can be used in any of the control modes.

When a parameter is selected — decay time, pre-delay, sweep speed, limits, etc. — adjustments are made by the Adjust/Select control, a digital slider that converts movement into value changes. The control operates like the sliders on a Lexicon 224X and

— SP2016 Remote Control —

needs to be centered before adjusting to the upper and lower limits. The slider has a centered detent and, as it moves to left or right, a "-" or "+" light comes on while the display window shows parameter changes. Moving the control back to center stops the adjustment. The Adjust/Select slider worked well most of the time for adjusting parameters that have a relatively finite number of variables, such as those in the reverb programs. It was also good for scrolling through *Command* or sequences. I found it harder to deal with the selection of delay times, however, when the slider moved in 0.1-millisecond increments from zero to 40 or even zero to 3276.5 milliseconds. For these parameters, the adjustment seemed to move either too fast or too slow, and it took time jockeying between the "+" and "-" zones to dial in on the desired value.

Besides being responsible for adjusting control modes, the slider also selects the different *Soft Key* functions that execute various program-dependent commands. Functions are displayed by scrolling through the Command key until a flashing red arrow appears next to the Soft Key, at which point the function may be executed by hitting the Soft Key. Alternative functions can be called up with the Adjust/Select slider, and initiated by hitting the Execute key. The primary Soft Key function in any program can be executed, even though not displayed, by simply hitting the key. This can act as a safety measure on some programs, such as those employing feedback, to prevent runaways. It also handles punch-in/out for loop edit, enable or disable inputs for reverb, plus many other functions.

The Command Key calls up various hardware and software functions to the display window. The functions include Soft Key (as discussed above), as well as System Help, Save User Preset, Kill User Preset, Line In/Out, Bus Address, Get Last Program, Self Test, and Configuration.

System Help is a description of front-panel controls that scrolls through the display window, and serves as a simplified mini-manual that could get a non-user off the ground. (One should read the manual, however). *Save* and *Kill User Preset* are fairly self explanatory; 65 user-modified programs can be stored and given names and numbers to be displayed in the window for easy recognition. The presets are easily eliminated in order to make more room in memory.

Line In/Out allows you to bypass the processor without turning it off, and is another convenience when tracing problems.

Bus Address allows the user to change the system address when taking commands from a computer, as in the case of operating multiple processors.

Get Last Program allows you to call back the previously executed program with all modified parameters. In this manner you can switch instantly between two programs during a mix.

There is a *Short Test* and *Continuous Test* program that can be called up when the special diagnosis ROM is inserted; more on this later.

Configuration simply displays the serial number, software configuration, and birthday of your processor. This completes the description of the Processor Control Block whose function, except for *Define*, are duplicated

SUMMARY OF EVENTIDE SP2016 GENERATOR II EFFECTS PROCESSOR/REVERB

Number of Inputs: Two XLRs, 10 kohms, balanced; range is -10 dBm to +24 dBm.
Number of Outputs: Two XLRs, 150 ohms, balanced, maximum output is +18 dBm, suitable for driving 600 Ohms or higher line-level impedences.
Frequency Response: 20 Hz to 16 kHz for most programs, 20 Hz to 8 kHz for programs that require extended memory.
Dynamic Range: 82 dB, typical.
Distortion: THD at 1 kHz is equal or less than 0.1%; 0.05%, typical; 0.1%, maximum.
Program Capacity: 11 Program ROM capacity at any one time. Each ROM can hold from one to five programs depending on memory required for each program. Any number of programs can be stored outside the machine and inserted in moments.
Presets: Up to 65 sets or program parameters can be stored in non-volatile memory.
Reverberation Time: Zero seconds to hours (program dependent); zero to 10 seconds, typical.
Frequency Contour: Program dependent; cut and boost at selectable low frequencies and high frequencies.
Pre Delay: Zero to 999.9 milliseconds (program dependent); zero to 400 milliseconds, typical.
Room Position: Program dependent; controls releative strengths of "reflections"; controls apparent room position of listener in relation to sound source.
Display: Two, 10 LED bargraph type meters for monitoring inputs or outputs. Program status displays (i.e. mono, 16 kHz). Main display is 16 alphanumeric characters.
Remote: Optional mini remote comes with 20-foot cable and DIN-type connector for immediate use. Remote box is 3½ by 1½ by one-inch (H × W × D), and has all processor controls except Define key and Display.
Standard Interface: IEEE-488 connector for computer control.
Additional Controls: Two Input faders, two Output faders, two Dry/Effect faders, Program Select, Parameter Select, Command key (for addressing presets, help text, self-diagnostic test), Execute key, Adjust/Select slider, Define key, and Soft key (executes special command features), Output Monitors button, and on/off switch.
Diagnostic Programs: Standard short and continuous self test.
Price: $9,495, including Generation II software.

Eventide Incorporated, One Alsan Way, Little Ferry, NJ 07643. (201) 641-1200.

□□□

For additional information circle #112

on the remote control.

The remote control is a palm-sized unit that attaches to the processor via a 20-foot cord. The SP2016's remote control is particularly useful, since it allows you to sit at the console while modifying programs, and maintaining a goodly distance from the processor itself. And you need to be as far from the unit as possible because the cooling fan — and this is no exaggeration — is louder than my refrigerator's motor! I found it quite annoying in the studio, and I'm *sure* Eventide could buy a quieter fan. But, even though you have a 20-foot cord on the remote, because of the narrow display-window height, it is uncomfortable to keep the unit more than six to eight feet away without developing severe eyestrain when modifying prorams. After 10 to 12 hours in the control room with smoke and fatigue, I found myself leaning in to get a good look at the display.

Computer-control Interfaces

Of course those of you with a computer (even an inexpensive one) can solve the noise and display problem instantly. Through the back-panel IEEE-488 interface bus, you can hook up most Hewlett Packard, Commodore Pets (except VIC), Apple II and III Series, Tandy/Radio Shack TRS-80 Model 1 and 3, or IBM PC and compatibles. H-P computers hook up directly, while others require special interface cards and/or cables. The computer will access all controls of the Processor Control block, allowing you to sit at the console with your keyboard and VDU while the SP2016 sits in another room or back in a sound-absorbent area. With suitable software, the VDU will display all controls and display-window descriptions. You're going to have to spend some time writing a program to give you total control, but I would venture a guess that Eventide would be happy to network those interested owners for cooperative efforts. (Hmmm. . . I wonder if this could be hooked up to an SSL?)

Effects Programs

The true test of any effects unit is the available programs and their quality. Engineers today are demanding more versatility in reverb and effects, many times choosing versatility over sound quality. Eventide is striving to provide the user with the most in programmability, while maintaining a high standard of quality. The SP2016 comes with 19 factory-loaded programs, each with a variety of user-adjustable parameters. Because of space limitations I cannot give full program descriptions, but briefly will touch on some while looking more closely at the most popular program types.

There are three reverb programs: *Room, Stereo Room,* and *Generic.* The reverberation patterns generally had a good clean sound with pretty smooth tails, and were capable of handling fairly substantial input levels without glitching. I personally thought they sounded a bit "hard" in the mid-range with a little too much over-ring on vocals. (The factory later told me that a European stone castle influenced the design of these programs, so that is probably what I am hearing). I laid down keyboards and guitars wet to track with these programs, and was quite pleased with the results.

Two plate programs are available: *Plate Reverb,* and *Hi Density Plate.* Both are mono-input/dual-output simulating dissimilar pickup position. The Plate works as you would except (I utilized it mostly for vocals), while the Hi Density Plate simulates an extremely dense plate that would carry very long decay times. Both effects are useful, with the latter more so for exaggerated effects.

It's becoming the "in thing" to digitally sample a sound, and then key it back into the track. *Loop Edit* allows this at be done with some nice variations. The captured signal may be edited on both the front and back segments, while still retaining the full loop recording. You can play back the loop or edited section continuously, or trigger it from the CV input or via the Soft Key control. You can also change pitch up or down by recording at one bandwidth and playing back at another, a trick that will lengthen or shorten your loop sample by a factor of two. At a 16-kHz bandwidth, maximum sample length is 1.636 seconds — I would like to see this extended, so that one could sample more than just drums, or short bursts of sound. It would be great to be able to put whole phrases or lines in memory but, as it stands, this program works well and is fun to use.

There are six various delay programs that should cover all the bases for you:

Dual Delay: Two independent, 16 kHz bandwidth lines with up to 800 milliseconds of delay, and the capability for variable-length infinite repeat.

Long Delay: Same function as above, except it is mono-input/dual-output, with up to 3.276 seconds delay at 8 kHz bandwidth.

Band Delay: This mono-input/dual-output delay allows the signal to be divided into four separate frequency

SUMMARY OF SELECTED REVERB AND EFFECTS PROGRAMS

GENERIC REVERB: *Factory program number — F26 Stereo/Mono*

Generic Reverb is designed to simulate the characteristics of many "standard" type of digital reverb algorithms, and it does accomplish this goal.

PARAMETERS	RANGE
1. Decay Time	(0.1 to 10 seconds)
2. Room Position	(Front to Back)
3. Room Size	(1/Small to 10/Large)
4. Pre Delay	(1 to 100 mS)
5. Low Factor	(-8 to +4)
6. Low Rolloff	(50 to 500 Hz)
7. High Factor	(-8 to 0)
8. High Rolloff	(1 kHz to 8 kHz)

Softkey: Disable/Enable Input Clear Resume Reverb

MULTITAP DELAY: *Factory program number — F24 Stereo/Mono*

PARAMETER	RANGE
1. Delay Taps	(1 to 50/Mono; 1 to 25/Stereo)
2. Last Tap	(10 to 120 mS/Mono; 10 to 600 mS/Stereo)
3. Envelope Shape	(Exponen Decr, Linearly Decr, Flat, Linearly Incr, Exponen Incr Triangular)
4. Type of Spacing	(Decr 1 to 5, Constant, Incr 1 to 5)
5. Output Mode	(Normal or Mixed)
6. Diff Delay*	(Output Tracked or Output Skewed)
7. Pre Delay 1	(0 to 400 mS)
8. Pre Delay 2*	(0 to 400 mS)
9. Feedback	(-99.9% to +99.9%)
10. FeedbackTap	(10 to 1600 mS/Mono; 10 to 800 mS/Stereo)
11. Signal Fed Back	(Single Tap, Multitap, Crossfed*)
12. Input Mode	(Mono or Stereo)

*Appears in Stereo Input Mode only.

Softkey: Zero Feedback

STEREO ROOM: *Factory program number — F08 Stereo/Mono*

The Stereo Room reverb program creates the ambience of a large concert hall, and offers clear and natural reverberation with nine variable parameters:

PARAMETERS	RANGE
1. Decay Time	(0.2 to 30 seconds)
2. Room Postion	(Front to Rear)
3. Pre Delay	(1 to 250 mS)
4. Low Factor	(-8 to +4)
5. Low Cut-off	(50 to 500 Hz)
6. High Factor	(-8 to 0)
7. High Cut-off	(1 kHz to 8 kHz)
8. Diffusion	(High, Med, Low)
9. Input Mode	(Stereo Chn. 1 Mono)

Softkey: Disable/Enable Input Clear Resume Reverb

Loop Edit: *Factory program number — F14*

PARAMETER	RANGE
1. Length	(20 to 1636 mS/40 to 3,272 mS)
2. Position	(0 to 1636 mS/0 to 3,272 mS)
3. Record Mode	(Punch In, Punch Out)
4. Playback Mode	(Control Mntr/1 Shot, One Shot)
5. Envelope Trigger	(On; Off)
6. Length Bandwidth	(Normal/16 kHz; Extended/8 kHz)

Softkeys: Punch In/Out; Release Loop; Playback Trigger.

HIGH DENSITY PLATE: *Factory program number — F29 Mono*

PARAMETER	RANGE
1. Decay Time	(0.2 too 100 seconds)
2. Pre Delay	(0 to 999.9 seconds)

Softkey: Disable/Enable Input; Delay Taps bands, delayed, staggered and resequenced if desired.

The *Multitap Delay* program can be thought of as a simulated tape recorder with playback heads whose positions, playback volumes and feedback configurations, are user adjustable. Your program material can run through this series of "playback heads," called *Taps*, in either mono or stereo. This powerful program is particularly useful for creating unnatural reverberation effects. In fact, a very effective "backwards" reverb effect can also be achieved — you can literally hum a simple melody into this program, and end up with it sounding like a 150-person chorus singing Gregorian Chants in the Notre Dame cathedral.

Long Digiplex: A three-tap variable-delay with 3.2765 seconds available at 8 kHz. It is mono-in/dual-out with feedback functions.

Dual Digiplex: Same as above except it offers 800 milliseconds of 16-kHz delay with stereo inputs.

The rest of the programs I'll gather into the special-effects group:

Time Scramble: Breaks up the input signal into segments, and scrambles them in the time domain in various ways.

Flanger: Extremely versatile in its parameters; if you don't know what this does, you're in the wrong business!

Envelope Flanger: Ditto with a follow mode.

Chorus: Adds two, four, six or eight voices to the input signal. Voices can be varied in placement around the listener by use of diffusion and feedback.

Musical Combs: This creates two sets of comb filters. The center frequency of the comb may be tuned over a two-octave range, and the musical interval can also be set. This program is worth some time investment for creating unique effects.

Lossless Room: Simulates a room of infinite reverb with parameters of high-frequency absorption, and decay rate variable in milliseconds, 1 second, seconds, minutes, or hours. All I can say is load it, stack it, and trigger it — you'll like it!

Dual Robots: Variable taps and spacing create robot-like voices. When I first heard this I thought it was really stupid. As a joke I put it on the lap steel guitar, but got a fantastic sound like someone playing through a carbon mike, and coming out of through an old-time radio.

Of course, all these programs can be manipulated and stored in any of 65 user memories. It will take time to familiarize yourself with the various functions, but it's worth the investment; there are lots of good sounds to tap (no pun). As new programs are created, the ROMs can be easily replaced by removing the top panel and inserting a new program into the zero-insertion sockets.

Internal Construction

With all this versatility, you can probably guess how many millions of chips are packed under that hood. (You guys in maintenance know, don't you?) Well, Eventide was really using their kidneys when the SP2016 was constructed, because it looks like a maintenance dream. When powered up it runs a little diagnostic on itself. By inserting the special diagnostic chip that comes with each unit you can run a short, more extensive self-test at any time, or a continuous diagnostic to look for bit errors. The latter allows you to run selected tests (up to 19 different tests), or skip any you desire. If there are problems, the display window will tell you what is wrong, and the location of any bad chips. Example messages could be: "low battery"; "BAD-15v power supply"; or "BAD IC 27/PC24" — the last referring to location #24 on the printed-circuit board. Each pass of the continuous test takes only 2¼ minutes, so you don't have to wait long to find out what's cooking.

When you take off the top by removing three screws at the front of the main PC board, the entire assembly lifts up like a car. It's neat and clean and very easy to work on. The combination of the self tests and physical layout means that there will be very little downtime lost to this piece of gear.

Eventide is shooting for longevity with the SP2016; it has designed the system with updates in mind, and has prepared well for the future. While there are some features and, even some programs that I think need some changes, my overall impression is favorable. The front-panel layout is clear and easy to operate. The output is quiet, even when at maximum level, and the input is capable of handling a lot of level, so you should have no problems with glitches. Most of the programs have a great deal of flexibility in their parameters, so you should be able to really fine tune your own sounds. If you are planning to buy a digital reverb, it is important to consider expandability and the future in relation to what is trendy today.

In a subsequent issue of *R-e/p*, I will explore the Eventide SPUD (Signal Processor User Development) system, which allows SP2016 owners to create their own programs and burn their own EPROMS.

I would like to thank Studio D in Sausalito, CA, for donating the session time necessary for me to make this assessment.

■■■

New Products

NEW Q.LOCK 4.10 SYNCHRONIZER AND ECLIPSE EDITOR FROM AUDIO KINETICS

Q.Lock's computer frame has been redesigned to accept four complete machine interfaces. The standard controller and its associated software will still synchronize three machines, but the operator now has instant access to any three of a "pool" of four machines, without PCB substitution. Provision has also been made in the frame for insertion of an optional VITC reader, with rear-panel BNC connections. An additional spare card position has been provided for the addition of an accessory card, such as the 16-Event Relay Card option.

•The Eclipse Editor is an intelligent controller that has been designed in conjunction with the Q.Lock 4.10 system outlined above, to offer full four-machine capability. The unit uses a high-definition, mini-VDU display, capable of showing 20 lines of information such as machine status, timecodes, set-up menus and other text.

A new Machine Access feature allows any machine to be accessed individually while the group remains in a synchronized state. Operations may be performed on this machine without affecting the rest of the machines status. Also a Machine Group Setup allows two types of Machine Group to be set up, each with a different machine as Master. Selection of which machine configuration is to be used is accomplished by pressing one key.

The system's operational parameters and general facilities may be set up from the keyboard, such as code standard, facility options etc. A number of assignable keys are provided for operators to program their own sequence of multikey operations, in addition to the many dedicated function keys provided. Also, the number of loop memories available for specialized use has been increased to 100.

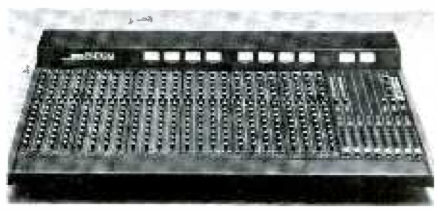
Eclipse is described as the first stage in an expanding Machine Management System that will provide multimachine control.

AUDIO KINETICS, INC.

For additional information circle #114

YAMAHA MC SERIES MONITOR CONSOLES

The MC1608M and MC2408M consoles feature XLR-type connectors on all primary balanced inputs and outputs. The MC1608M has 16 input channels, and the MC2408M 24. Other than this, both consoles have the same basic features, with 10 VU meters, (eight master busses, plus echo #1 and #2) eight master outputs, as well as two auxiliary sends and two fully assignable auxiliary returns.



The consoles are modular in construction, with blocks of four input channels for easy service when necessary. Each input channel features a pad switch and gain control with peak LED; phase reversal switch; three-band EQ with sweep midrange; two post-EQ and pre-fader auxiliary sends; eight rotary master send controls; channel on/off and cue switches with input channel cue priority. With this latter feature, when an input channel cue switch is pressed the previous master cue is cancelled.

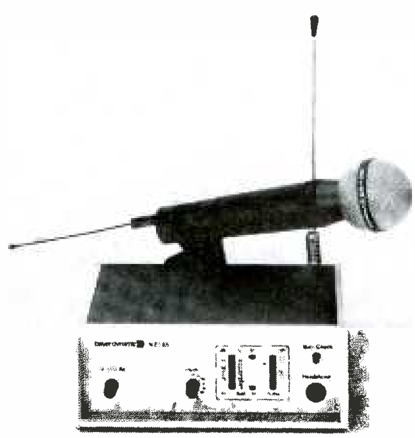
Suggested retail price of the MC1608M is $2,895, and $3,995 for the MC2408M.

YAMAHA INTERNATIONAL CORPORATION

For additional information circle #115

BEYER DEBUTS MODEL S185US WIRELESS MICROPHONE

The new hand-held wireless microphone system operates in the 174 to 216 MHz range, and features a switchable built-in limiter and low-noise compander system.



The S185US transmitter can be used with the BM85 ribbon capsule, or the EM85 cardioid condenser capsule. The CV85 adapter, when used in conjunction with the transmitter, allows any "studio-quality" condenser capsule in Beyer's MCM Series to be used in any wireless application — a configuration that makes possible a hand-held wireless long or short shotgun microphone.

The Model TS185US pocket transmitter offers all the features of the hand-held model in a considerably smaller size. The ability to interface directly with Beyer's MCE5 lavalier condenser makes the TS185US ideal for live-concert applications, where wireless systems have to go unseen.

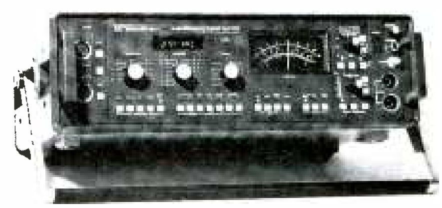
The NE185US wireless receiver includes a low-noise compander system, a built-in adjustable electronic squelch, switchable output level, line-level or variable-level and two LED indicators for field strengths and audio level. AC and DC powered (external battery supply), the unit is available in either single- or three-channel versions; both are available in a diversity configuration.

BEYER DYNAMIC, INC.

For additional information circle #116

NEW MICROPROCESSOR-CONTROLLED AUDIO TEST/MEASUREMENT SYSTEM FROM TECPRO

The MJS401D Audio Test Measurement System features one-button set-up, and "intelligent" interlocking for speed and accuracy. An IEEE bus allows for optional computer control.



Single- or twin-output plug-in oscillators, which employ 6802 microprocessors to interface with the micro-based MJS401D, are available as accessories. The basic system is priced at $3,999; single and twin oscillators are priced at $962.50 and $1,181.25 respectively.

TECPRO, INC.

For additional information circle #117

TANDBERG UNVEILS TCD 900 SERIES CASSETTE DECKS

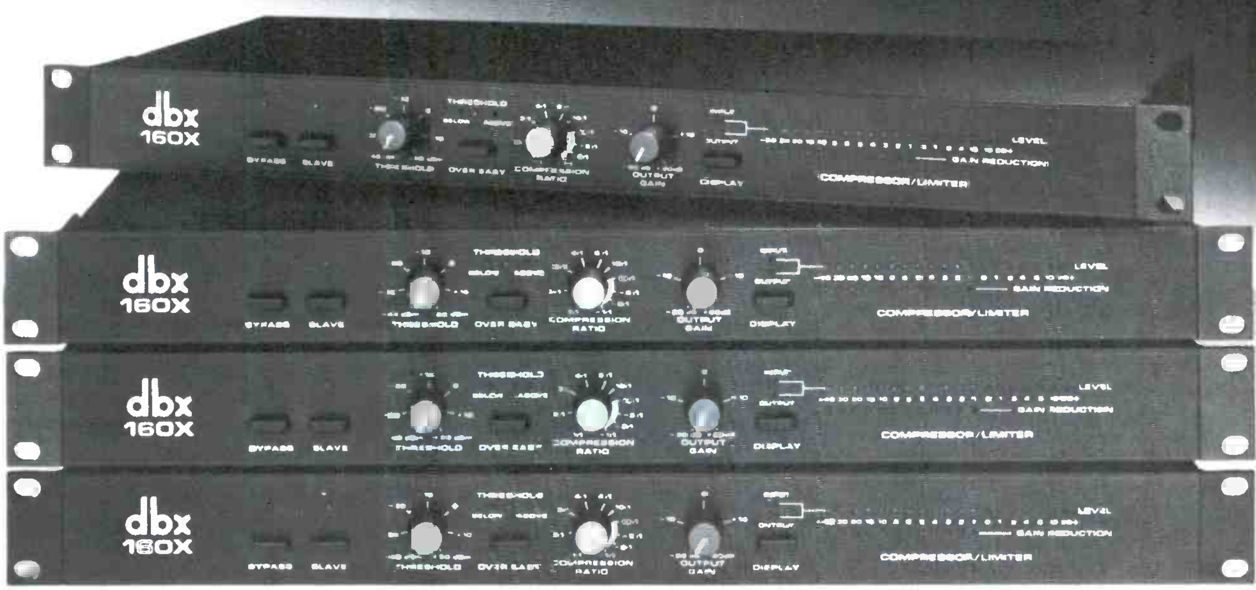
"Tandberg's new Series TCD 900 is a superior and cost-efficient alternative to the practice of using inferior home tape decks for professional applications," says Peter Wellikoff. "These new professional cassette decks offer unparalled sound capability, advanced mechanical and electronic design, plus extraordinary control flexibility based on an eight-bit microprocessor with 32K or EPROM memory.

The new Series incorporates a discrete three-head system; dual-capstan, closed-loop drive; four servo-control, belt-isolated motors; peak-reading, equalized meters (which respond to a two-millisecond peak within 1 dB); built-in oscillators for bias, record current, and azimuth adjustment; plus front-panel playback azimuth adjust-dbx

We listened to you— now we'd like you to listen to us

The dbx 160 was one of the world's best-selling compressors, but there were still a few areas you wanted improved: stereo linking, easy rack-mounting, clearer metering, selectable compression.

We listened. The result is the 160X. Visit your authorized dbx professional products dealer and listen for yourself.



dbx Inc. Professional Products Division 71 Chapel Street, Newton, Mass. 02195 U.S.A.
Telephone (617) 964-3210 Telex: 92-2522

For additional information circle #118

ment and pitch control. Dolby B and C noise reduction systems also are included.

A built-in microprocessor operates the LED real-time counter in minutes and seconds, as well as the units' auto search mode, recap function, and, in the case of the TCD 910, the four-seconds auto record mute (with manual override). The microprocessor also features a volatile memory that can store as many as 10 cue points. Serial interface for computer operation is made possible with an optional RS-232C port. "With the computer-interface function," explains Wellikoff, "these decks can replace cart machines in the studio."

TANDBERG OF AMERICA, INC.

For additional information circle #119

SCV MODEL FA4-2 STEREO CROSSOVER

The Model FA4-2 is an active stereo three- or four-way crossover designed into a 19-inch rack mountable package utilizing only one rack space. In addition, the crossover uses the Butterworth-Linkwitz transfer function giving 24 dB per octave slopes. Crossover frequencies are fixed via plug-in filter modules, of which 10 different modules are available between 10 Hz and 10 kHz. A blank module is also available for any factory non-standard frequency.

The unit features Lo, Mid, Hi and Ultra-Hi level controls, with LED level indicators, delayed turn-on protection relay, and a three/four-way switch per channel.

SCV, INC.

For additional information circle #120

J.L. COOPER UNVEILS RANGE OF MIDI CONTROLLERS

•The MIDI Oberface allows an Oberheim DSX sequencer to interface with other MIDI equipment. All operations of the DSX remain strickly stock, and the CVs and Gates still operate in the same way. Note storage is broken down into two groups: Main notes and Auxiliary notes. Main notes would play back on MIDI channel #1; auxiliary notes on #2. Either or both of these groups may be split into two channels of four voices. Note On and Off and program change commands are supported; Velocity and Pitch Bend are not. Suggested retail price is $495.

•The MIDI Switch Box I is a rack-mountable unit that allows the attachment of up to eight sources and up to 10 destinations. Each destination has a selector knob on the front panel that chooses the MIDI source (A-G). Suggested retail price is $395.

•The MIDI Switch Box II can accommodate eight sources in, and up to 16 sources out. The unit is microprocessor controlled, and can be instructed to change the connection configuration from a remote MIDI source. The configuration status is displayed via 7-segment LEDs on the front panel. The unit has an internal battery backed-up memory, and may store up to 16 different patch configuration. Suggested retail price is $1,195.

•The MIDI Switch Box III allows two sources in, and up to four sources out. It is also programmable and can remember two different configurations. By pressing the built-in footswitch, the unit goes back and forth between the two programs. LEDs indicate program A and B. Suggested retail price is $220.

•The MIDI Switch Box IV is simply a MIDI on/off box. A foot-operated switch toggles the connection between the MIDI In and MIDI Out on or off; an LED indicates when the MIDI slave is on. Suggested retail price is $75.

J.L. COOPER ELECTRONICS

For additional information circle #121

TOA ELECTRONIC MUSIC MIXING SYSTEM

The four-channel mixing system, expandable to 10 channels, is designed primarily for keyboard and guitar synthesizers, and drum machines. The system consists of the rack-mounting D-4 and the D-4E Expander Unit, and is intended for live sound applications (on stage stereo and mono mix) and for studio recording applications.

Configured as 4 × 4 × 1 (sum out), the D-4 connects directly to its companion, the D-4E, thereby expanding the system to 10-channel capability. The compact, self-powered system offers both +4 dB and –10 dB outputs for connections to either tape machines or power amplifiers. Accessory patches on the stereo left and right busses offer added flexibility.

Each input channel features a ¼-inch jack input, RCA jack input, two-band EQ, trim control with LED peak indicator, post-effects send, aux send, pan control (to L and R mixing busses), and level control.

In addition to its phone jack inputs, the fourth input channel of the D-4 provides an electronically-balanced, three-pin XLR mike input with switchable 48-volt phantom power. The D-4E provides similar XLR inputs for channels five through eight. All patch point inputs are selectable to pre- or post-EQ and fader, for use with signal processing equipment. Each D-4/D-4E input has a direct output, which facilitates multitrack recording applications.

Together, the D-4 and D-4E offer first-generation MIDI capability with a one-by-eight MIDI-THRU function, which enables control of multiple synthesizers with a single MIDI-IN signal.

The four-channel D-4 and the six-channel D-4E each have a list price of $499.

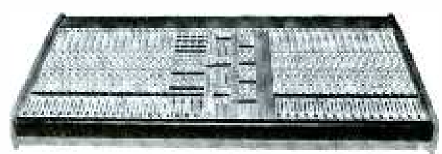
TOA ELECTRONICS, INC.

For additional information circle #122

HARRISON INTRODUCES HM-4 AND SM-4 SOUND REINFORCEMENT CONSOLES

The HM-4 (house main) and SM-4 (stage monitoring) systems are medium-scale products derived from the HM-5 and SM-5 touring consoles. The HM-5 and SM-5 were developed for Showco and Clair Brothers, under a special product development agreement with Harrison Systems, Inc.

The SM-4 and HM-4 systems are intended to allow a wide variety of sound reinforcement clients to take advantage of the features and facilities previously available only on the HM-5 and SM-5 systems, such as four-band parametric EQ, VCA grouping, and an eight-group mute matrix.



The HM-4 system includes eight mixable auxiliary sends, four stereo audio groups, redundant main stereo outputs, and a four by four stereo group matrix. The SM-4 system includes eight mix sends plus eight additional fully matrixed mix outputs from four mono audio groups.

Both consoles will be available in either fixed theatre or portable touring mainframes, which will include XLR-type mike inputs, and 30-pole DIN-standard Tuchel connector interfaces for patch systems.

HARRISON SYSTEMS, INC.

For additional information circle #123

INTERSONICS MODELS 2-15 AND 2-18 SUBWOOFERS

Latest in the generation of SDL IM (servo-drive loudspeaker) systems are the 2-15 and 2-18 subwoofers, which employ the same drive systems incorporated in the company's larger TPL Series of subwoofers. The driving force is supplied by a special motor directed to a pair of specially designed, long-excursion radiators.

The SDL system is said to permit outstanding acoustic performance, low harmonic distortion, and usually good waveform accuracy. This, combined with the large excursion capacity in excess of one-inch peak-to-peak, permits a revolutionary level of performance in a small enclosure, with deep, powerful, and clean sound.

SOUND PHYSICS LABS, DIV. OF INTERSONICS, INCORPORATED

For additional information circle #124

New Products

NEW PRODUCER SERIES FOUR-TRACK CASSETTE SYSTEM FROM YAMAHA

The new system includes the MT44D four-channel cassette recorder, the lightweight RM602 six-in, two-out mixer, and the RB35B rack and patchbay. Also available are two rack-mount peripherals: the GC2020 compressor limiter; and GQ1031 one-third octave graphic equalizer.

Heart of the system is the new MT44D four-track cassette recorder that allows up to four tracks to be recorded simultaneously with Dolby B- or C-type noise reduction. Controls include full logic transport with fast forward and reverse cuing, plus autolocator functions. Each of the MT44D's four channels has a fast-response LED level indicator, while a jack on the rear panel provides for hands-free punch-in when used with a standard footswitch. As with its predecessor, the MT44, the new deck can operate in either two- or four-channel mode. Suggested list price of the MT44D is $535.

•The RM602 is a compact, lightweight, six-in/two-out mixer with four input channels that double as tape playback inputs. (The other two channels have RIAA inputs for any standard phono cartridge.) Each input channel has two-band EQ, pan, effects and stereo monitor sends. Suggested retail price is $395.

•The RB35B patch bay and rack assembly, designed exclusively for the MT44D and RM602, enables the Personal Studio System to be integrated into a single compact unit. Suggested retail price of the RB35B is $165. The entire Personal Studio System — MT44D, RM602 and RB35D — has a $1,095 suggested retail price.

•The GC2020 is a two-channel compressor-limiter with an expander-type noise gate to further aid in noise reduction, while the GQ1031 provides 31 bands of equalization. Suggested retail prices are $295 and $245, respectively.

YAMAHA INTERNATIONAL CORPORATION

For additional information circle #126

BOULDER 500 POWER AMPLIFIER ADAPTED FROM 990 OP-AMP

Deane Jensen's 990 op-amp has provided a circuit-theory basis for the new power amp. Ultra-low THD and elimination of transient distortion are said to be inherent in this design, resulting in outstanding sonic clarity.

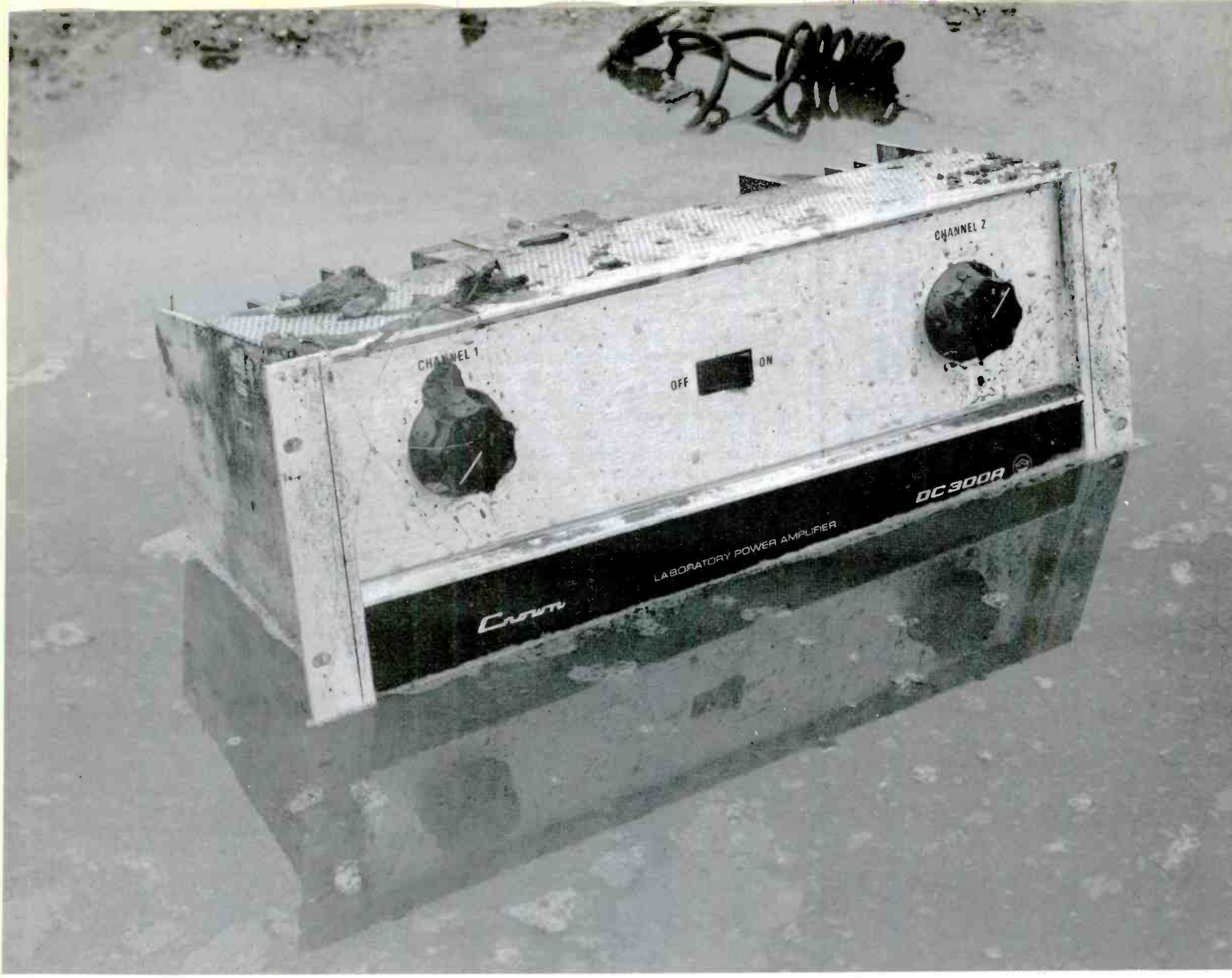
The Boulder 500 delivers continuous power of 150 watts stereo and 500 watts mono into 8 ohms, or 250 watts stereo into 4 ohms. Two sequential stages help yield low THD of 0.0015% across 20Hz to 2 kHz, rising to 0.005% at 20 kHz. High power bursts in excess of 1,300 watts mono are possible into very low impedance loads, such as dual-woofer monitor systems. Excellent control over reactive loads is maintained by supplying heavy currents during opposite (to the current) voltage polarity, the company says.

Twenty-eight heavy-duty, 250-watt, metal-output transistors on massive side heatsinks are said to rarely require limiting of any kind for real-world signals. Linear temperature tracking protection circuits dynamically monitor the amplifier's output voltage and current, and the polarity of each. Audio quality will not be degraded until the safe operating limit is approached regardless of load ortemperature.

Suggested retail price of the Boulder 500 is $2,450.

SILVER LAKE RESEARCH

For additional information circle #127



In the early evening of Sept. 17, 1973, Jay Barth was at the wheel of a 22 ft. utility truck that was loaded with sound equipment. Just south of Benton Harbor, MI an oncoming car crossed the center-line; fortunately Jay steered clear of the impending head-on collision. Unfortunately, a soft shoulder caused the truck to roll two and one half times. Exit several Crown DC-300A's through the metal roof of the truck's cargo area.

The airborne 300A's finally came to rest — scattered about in a muddy field, where they remained partially submerged for four and a half hours.

Jay miraculously escaped injury; the amplifiers apparently had not.

Unbelievably, after a short time under a blow-dryer all the amps worked perfectly and are still going strong.

The rest — and the truck, is history.

crown
1718 W. Mishawaka Road, Elkhart, IN 46517
(219) 294-5571

For additional information circle #129

New Products

SYSTEM 2.1 MONITOR SPEAKERS FROM KLARK-TEKNIK

The Klark Acoustic System 2.1 is described as a fully complementary integrated design, optimally matching the built-in electronic components such as amplifiers and compensating crossover networks with the electro-acoustic parts of the system.

The two-way bass reflex system with unique frontal raised sound board profile is said to promote a stable and smooth off-axis dispersion pattern, emphasizing accurate stereo location and placement of sounds desirable when building up a three-dimensional sound landscape. Complementary MOSFET amplifiers are located in a screened rear cabinet compartment.

To allow easy installation and quick operation of the system, fully adjustable mounting trolleys have been designed, enabling optimum positioning and thus freeing the user from physical placement inflexibility, therefore resulting in more accurate reproduction by overcoming distracting phase- and time-based problems often encountered with console-mounted speakers.

A custom-designed LF driver features non-resonant Neoflex cone material and "edge wound" copper strip wire on large heat dissipating Nomex coil former. An inverted-dome HF driver combines rigid non-flex woven fibre glass with semi-hemisphere bonded aluminium voice coil.

Recommended pro-user price is $1,650, including stand.

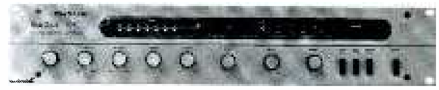
KLARK-TEKNIK ELECTRONICS, INC.

For additional information circle #130

URSA MAJOR STARGATE 626 DIGITAL REVERB AND EFFECTS UNIT

The StarGate 626 is an enhanced version of the Model 323 digital reverberator, and adds eight new reverb and effects programs to the original unit's eight "rooms." The 626 also utilizes the latest 256K RAMs for expanded memory/time delay.

At present, the 626's 16 "rooms" include:
•*ROOM 0:* A fast-diffusing plate type program with no discrete pre-echoes;
•*ROOMS 1 thru 8:* The eight reverb rooms of the StarGate 323, with no changes;
•*ROOMS 9 and A:* Long-space reverb rooms with smooth decays up to 15 and 20 seconds, respectively;



•*ROOM B:* "Reverse reverb" based on Room #3, but with echoes rising in gain to achieve a reverse effect;
•*ROOM C:* DOL with full bandwidth and adjustable delay time and feedback gain;
•*ROOM D:* A DOL similar to Room C, but with a stereo audition delay pickup pattern;
•*ROOM E:* A DOL similar to Room C, except that the left channel taps are set at one-half the displayed value. The right channel taps are set at the displayed value, as are the feedback tape — the effect is a decaying feedback heard alternately left and right;
•*ROOM F:* "Dual Echo" effect uses a stereo audition delay pickup pattern, as in Room D. One feedback tape is placed at the displayed value, and another at a fraction of the displayed value — the resulting feedback pattern begins as a distinct echo and gradually blurs into a reverb-like sound.

The 626 can be wired for external control of a Freeze function that locks up memory to repeatedly replay the last two seconds of sound stored before freeze was activated. The frozen sound can either be reverb or pure delay; moreover, the full range of Rooms and Delays can be used to hear the frozen sound in a great variety of ways.

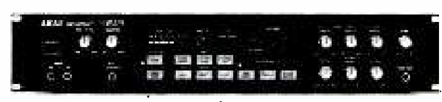
Pro-user price of the Stargate 626 is $2,500; upgrade kits to convert the 323 into the latest configuration will sell for $500.

URSA MAJOR, INC.

For additional information circle #131

AKAI INTRODUCES MODEL 612 SAMPLING UNIT

The new six-voice polyphonic Model 612 SAMPLER™ will interface with any MIDI keyboard to sample any digital or analog signal up to eight seconds. From the front-panel controls the user can loop the signal from any point for an unlimited playback time.



In just eight seconds, the new unit can faithfully duplicate any sound or preset from any MIDI-equipped synthesizer. Via MIDI, a sampled note can be played up and down any MIDI keyboard and, in addition, can be blended with other presets already in the synthesizer.

Suggested retail price of the Model 612 SAMPLER is $999.

INTERNATIONAL MUSIC COMPANY

For additional information circle #132

YAMAHA INTRODUCES S10X and S20X COMPACT PRO-AUDIO SPEAKERS

With a quoted frequency response of 65 Hz to 20 kHz, the S10X and the S20X can handle up to 75 watts and 150 watts RMS, respectively.

"While the S10X and S20X will not deliver the power needed for a concert, they are ideal for a number of smaller sound reinforcement and monitoring applications," states Yamaha Combo Products division manager, Bill Hinely. "They are also ideal where space is limited using the optional mike stand adapter, free angle clamp, and ceiling and wall brackets."

Both speakers feature a newly designed four-inch, full-range speaker that uses a specially formulated carbon-fiber cone. Not only is the carbon fiber lighter and much more durable than traditional paper cones, it is said to be extremely rigid, for precise response with minimal distortion.

The S10X, with one four-inch speaker, measures 9.5 by 6 by 6.5 inches, and weighs six pounds. Weighing 10 pounds the S20X measures $11\frac{1}{2}$ by $7\frac{1}{2}$ by $7\frac{3}{4}$ inches, and features two four-inch speakers.

Suggested retail price of the S10X is $120, and $180 for the S20X.

YAMAHA INTERNATIONAL CORPORATION

For additional information circle #134

SOLID STATE LOGIC ADDITIONS TO STUDIO COMPUTER SYSTEM

The SSL Integral Synchronizer and Master Transport Selector is a five-machine synchronizer that is said to combine the operational power and simplicity afforded by a central computer with the speed, versatility and reliability of fully distributed local processors. The system automatically calculates offsets and stores "Sync Presets" to reduce subsequent setup time.

A remote changeover function allows any one of three ATR, VTR or film machines to be designated as Master from the console, instantly switching all timecode, tach pulse, direction sense and transport logic lines to the central console controls.

The SSL Programmable Equalizer consists of a console-mounted control panel and a remote electronic package which interface with the Studio Computer. It provides two channels of three-band parametric EQ, and stereo panning. All functions are dynamically programmable, and the continuous sweep nature of the device is described as making it an ideal unit for simplifying dialog matching and EFX EQ and positioning.

SOLID STATE LOGIC

For additional information circle #135

SEQUENTIAL LAUNCHES MULTI-TRAK SYNTHESIZER

The new six-voice synthesizer can be used to play six completely different instrument sounds at the same time, in one of three ways. The first is by layering several sounds on top of each other, and then playing them from the keyboard. "SuperStak" mode enables the synth to play six sounds in unison; to play two keys at a time with three sounds on each; or to play three keys at a time with two sounds on each.

A second way of playing the Multi-Trak, as a multi-timbred "ensemble," is by recording different instrument parts using the synth's on-board digital recorder. Four separate sequence locations are provided, with a total memory capacity of 1,600 notes. Once recorded, sequences can be linked together in any order to create complex songs. A built-in metronome and five different resolutions of auto-correction are provided to correct any minor timing imperfections.

A third method for creating complex, multipart music is by combining the live playing

New Products

capabilities of the instrument with its multi-track recording feature. For instance, a bass drum part can be recorded on track #1, a snare drum on track #2, bass on track #3, and a brass line on track #4. The two remaining voices can be used during playback.

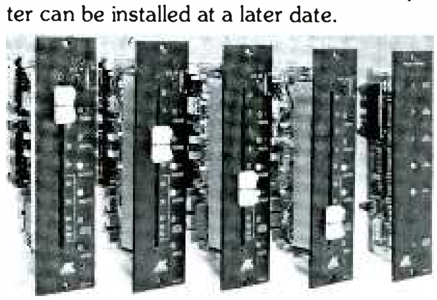
The Multi-Trak features a five-octave velocity keyboard that can be used to control loudness, brightness, or modulation amount. A built-in stereo chorus provides programmable on/off chorusing, as well as additional controls to vary rate and depth. Also featured include a latchable arpeggiator to latch or release the notes from the front panel, or with an optional foot switch, plus easy transpose and cassette dump.

SEQUENTIAL

For additional information circle #138

AUDIO KINETICS AUTOMATION FADER

In conjunction with the development of the MasterMix disk-based automation system, A-K has produced a VCA fader that can be used independently of the automation package. Although designed originally to enable non-automation-ready consoles to be automated, audio quality and specifications are described as being so good that the fader can be fitted to a console, together with only the automation interface; the result being a high-quality fader coupled with the unlimited subgrouping and solo functions offered by the automation system. The MasterMix computer can be installed at a later date.



Typical specifications for the fader are: noise less than -100 dBm; and distortion less than 0.01%. In addition, a simple push switch enables the VCA to be bypassed.

The fader can be supplied in various panel sizes to fit different consoles. Custom sizes and colors can be arranged.

AUDIO KINETICS, INC.

For additional information circle #139

AKG UNVEILS D321 DYNAMIC VOCAL MICROPHONE

Styled along the general D300 Series' lines, the latest addition is a dynamic hypercardioid vocal microphone said to be primarily characterized by extreme insensitivity to handling and pop noise.

The transducer is a fundamentally new dynamic design patented worldwide. The diaphragm is fixed directly to the transducer case, while the magnet is fixed to the case by an elastic element. Magnet and diaphragm are manually turned to the same resonance frequencies. Mechanical vibrations will excite both elements to vibrate in phase so that no electrical signal and, consequently, no noise is produced. Shock mounts around the transducer protect it from damage due to violent movements. The mike's compensation system also provides up to 20 dB better signal-to-noise ratio than current systems.

The unit's housing is thick-walled throughout to withstand both severe climatic (high and low temperature) and high mechanical stresses. The screw-on wiremesh cap is made from stainless steel, and the double windscreen beneath it is a fabric covered reticulated foam pad that effectively reduces wind and pop noise.

AKG ACOUSTICS, INC.

For additional information circle #140

UHER 160-AV PORTABLE STEREO CASSETTE RECORDER

With its unique combination of features, Uher says, the 160-AV is the most complete cassette recorder available in the world, and includes a DIN accessory facility for synchronized film dubbing. The unit can be powered by six dry cells, NiCad rechargeable batteries, 12-volt car batteries, and mains supplies.

The Model 160-AV also features both Dolby Type-B and -C noise reduction; three built-in speakers for on-site monitoring; separate right and left level controls and switchable automatic level control with two time constants, and twin peak-reading meters with dB scale.

The deck has a front-loading cassette compartment, three-way tape selector, line, mike, radio and phono inputs, as well as line, radio, monitor, headphone and speaker outputs.

Suggested retail price for the Uher 160-AV is $998.

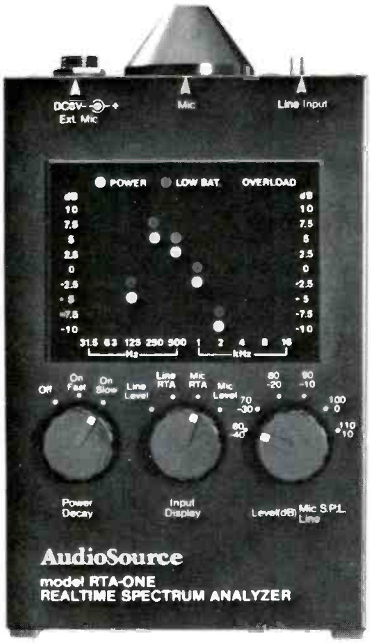
UHER OF AMERICA, INC.

For additional information circle #141

RTA-ONE REAL TIME ANALYZER FROM AUDIOSOURCE

Designed to withstand the rigors of daily use — not only indoors but sometimes in outdoor surroundings — the unit features a durable metal case that is said to render the RTA-ONE far more impact-proof than analyzers housed in plastic.

The device can be employed as a simple sound pressure level meter to gauge speaker output, or used to provide complex frequency response curve measurements. In addition to serving as a hand-held device, it can also be hardwired into a system for metering purposes.



The instrument divides the audio spectrum into 10 single-octave bands for instant visual representation of frequency response, an easy-to-read LED display, and a decay knob with both "fast" and "slow" settings. A level control adjusts the unit's sensitivity to correspond with the overall amplitude of sounds being monitored; this assures they are read out by the display, which covers a 20 dB span from 31.5 Hz to 16 kHz.

Also handy for the professional user is the RTA-ONE's protective hard leatherette carrying case with convenient handle and space for such optional accessories as AudioSource's RTA-ONE remote mike ($24.95 suggested list) and PNG-ONE pink noise generator ($44.95 suggested list). Power is provided by four AA batteries (not included) or an AC adaptor (available as an optional accessory for $12.95). The RTA-ONE carries a suggested retail price of $199.95.

AUDIOSOURCE

For additional information circle #142

SOUND WORKSHOP UPDATES SERIES 34 CONSOLE AND COMPUTER

The "B" revision of their Series 34 features a new modular patch bay system and a new console interface. The patch-bay system uses steel-framed TT-style jacks with a high-

strength subassembly, and connects to the console interface in the right console leg via computer-type wiring. Tape machines and other studio gear wire to the console interface via AMP multipin connectors.

Also featured is the new ARMS-II console computer, whose functions are sped up by a factor of two. The increased speed is accompanied by several new software-based features and functions. ARMS-II is available for retrofit into existing consoles, and may be specified for OEM installation from a variety of console manufacturers.

SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS, INC.

For additional information circle #143

LEXICON 1300-SERIES AUDIO-DELAY SYNCHRONIZERS NOW COMPATIBLE WITH TEKTRONIX 110-S VIA SERIAL CONTROL OPTION

A serial control option has been developed, that makes the stereo and mono versions of the Model 1300 Digital Audio Delay Synchronizer compatible with the Tektronix 110-S four-field frame synchronizer. The option, which is RS-422/RS-232 hardware-ready, is said to enable the Model 1300 to achieve extremely precise machine-to-machine synchronization for multiple-channel audio.

The new option, which provides coded delay times via serial ports, is the third in a series of options for the 1300; options that detect video-phase differences and delay-pulse widths were offered previously.

LEXICON, INC.

For additional information circle #144

For additional information circle #145

New Products

PEAVEY CL-1 LOUDSPEAKER SYSTEM

Designed for use as a small centrally-located cluster system optimized for reinforcing speech and music, the unit is said to be ideally suited as a permanent indoor sound system installation for small auditoriums, churches, and schools.

The CL-1 combines a multidriver array of six, six-inch diameter speakers in a closed-box design for the low/mid-frequency range, and 22A compression driver coupled to a CH-3 constant-directivity high-frequency horn.

The system is available in a black finish with a perforated metal grill frame (black) protecting the speaker array, or a white-oak veneer finish with a cloth-covered black grill frame for special purpose applications where aesthetic considerations must be satisfied.

Suggested retail price of the CL-1 OAK is $399.50; CL-1 (BK) $349.50.

PEAVEY ELECTRONICS

For additional information circle #146

SONY DMR-2000 DIGITAL AUDIO RECORDER

The two-channel U-Matic videocassette recorder is specifically designed for digital audio mastering purposes.

The DMR-2000 interfaces with the PCM-1610 processor, and has a built-in head cleaner that is automatically activated every five minutes to prevent any possibility of head clogging. It will also interface with the DAE-1100 digital audio editor via the IF-5850 interface unit.

SONY CORPORATION OF AMERICA

For additional information circle #147

REPLACEMENT MIKE BOOM LOCKSCREWS FROM BLACK AUDIO DEVICES

Since factory replacements are extremely hard to find, the replacement lockscrews offer an alternative to retiring an expensive boom from service, or finding a bolt and using pliers. In addition, the new range of lockscrews are described as being far easier than factory replacements to grip and tighten hard on the boom shaft.

Lockscrews have plastic molded handles on hard metal threads, and fit AKG-type booms in four styles and three colors.

BLACK AUDIO DEVICES

For additional information circle #148

360 SYSTEMS ANNOUNCES MIDI BASS

The new unit plays digitally recorded or sampled electric-bass and many other sounds, stored on plug-in memory chips. The Midi Bass can be played by any keyboard, guitar or sequencer having MIDI connectors. A single cable conects the two, and the Midi Bass is driven by the controlling instrument.

Since the unit's designers are themselves bass players, they realized more than a MIDI expander module was needed. The Midi Bass can be set to follow only one of a performer's hands, by programming a limit point within the keyboard range. And programmable key priorities are said to extract excellent bass lines by following only the lowest note, last note, or highest note played within a chord.

Touch-sensitive keyboards will fully con-

For additional information circle #149

trol the dynamics of the MIDI. A "Slapped" and "popped" funk bass is included in the basic unit, and three additional sounds can be added by the user at any time.

Current sounds in the library include Jazz basses with roundwound and half-round strings, P-Bass, Ripper, Alembic, Half-speed Guitars, Stand Up Bass, and various exotic sounds such as Clavinet, Guitar Chords in Open Fifths, and Best of Handclaps.

Suggested retail price of the MIDI Bass is $395; alternate sounds are $50 each.

360 SYSTEMS

For additional information circle #150

BGW SYSTEMS UPGRADES MODEL 85 POWER AMPLIFIER

The new Model 85 now delivers 35 watts per channel driving eight-ohm loads. Along with the increased power output, the new amps feature black anodized, brushed aluminum front panel and improved noise characteristics. New low-feedback discrete circuit design is said to result in exceptionally natural sound, coupled with the elimination of transient intermodulation distortion. A toroidal power transformer provides minimum size, weight, and low stray field and acoustic noise.

Three versions are available: the Model 85, which features single-ended inputs that accept ¼-inch phone jacks for unbalanced applications; Model 85-01, which features high-performance, active balanced input circuitry and XLR connectors, which allows BGW to guarantee a minimum of 70 dB common mode rejection; and the Model 85-06, which has built-in dual-input transformers that provide 15 kohm input impedance, and utilizes XLR connectors.

Other features include welded steel construction for maximum mechanical integrity and RFI shielding; modular construction to provide simplified servicing; a mono bridge switch to allow high-power, single-channel operation; transient-free circuitry to prevent speaker pops or extraneous noise; and detented front-panel gain controls and headphone jack.

Suggested retail price for the Model 85 is $449; Model 85-01 $499; and Model 85-06 $584.

BGW SYSTEMS, INC.

For additional information circle #151

GARFIELD ELECTRONICS TRIGGERING SYSTEM DRUM DOCTOR

From drum pads, pickups, or drum audio, each of the six channels provide rising and falling edge, five-volt triggers for Linn, Oberheim, Drumulator and MXR drum machines;

a dynamic trigger output for Simmons, Dynacord, TechStar and Fightman kits; dynamic MIDI triggers assignable to any note in any MIDI channel; and a dynamically controlled, gated voltage-controlled amplifier that adds dynamics to non-dynamic drum machines such as the LinnDrum, DMX and Drumulator.

The new unit is said to provide all formats and alternate MIDI modes, including program change from pad, sequencer advance, and legato triggering.

Suggested list price of the Drum Doctor is $1,495.

GARFIELD ELECTRONICS

For additional information circle #152

— *Classic Audio Consoles* —

Trident A Range VG 40/24/24 (ex Chateau, L.A.)	75k
Trident TSM VG 40/24/40 (ex Vineyard U.K.)	55k
Neve 8108 VG 56/48/56 (ex. Abbey Road, London)	130k
Neve 8108 Ex 48/48/48 Necam I	160K
Neve 8048 VG 30/16/24 1081 EQ (Polygon, France)	55k
Neve 8038 VG 36/16/24 1074 EQ Extra (ex Cresent, U.K.)	75k
Neve 8068 Mint 32/16/32 Refurbished	85k
Neve 8078 VG 36/16/24 Necam II	145k
MCI 636 VU Ex 28/24/28 Auto, 28 param.	32k
Harrison MR-2 Ex 48/32/48 Auto	75k
Quad Eight Coronado 36/24/36 Auto, discrete	35k
Soundcraft IIIB 32/24/24 4 Band EQ, TX less	22k
API/DeMedio VG 24/16/24 550 EQs	17k

-*Tape Transports*-

MCI JH 16/24T Loc III	17k
MCI JH 114/24T Loc III	18k
Ampex ATR 104	9.5k
Ampex ATR 102	6.5k
MCI JH 110 B 2T	4.2k
MCI JH 110B 4T	7K
Ampex MM 1200 New Head 24T+16T	20k
Otari MTR-90 II 24T/Loc	28k

— *Microphones and Processing Gear* —

3-C24 EX	2.2k
3-C12 Mint	2k
3-Neumann SM 69 VG	1.6k
4-Neumann U 67 VG	1.2k
2-Neumann M 49 VG	1.6k
10-Neumann KM 54	.2k
2-Neumann U47 Fet VG	.6k
2-Neumann M250 Ex	2k
Necam II 40 Channel retrofit Neve, Trident, API	40k
Neve EQ 4 Band 1095	1.2k
Neve EQ 3 Band 1064, 1073 EC	.75k
EMT 140 ST (Tube) Stereo	5.5k
EMT 250	18k
Lexicon 224 XL	9.5k
Eventide 1745M Pitch/Shift	1k
Dolby M24H	1.2k
Dolby 361 Mint	.85k
AOR Vocal Stresser #769XR/Limit/Exp./Gate w/EQ	1.1k

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For additional information circle #154

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Hollywood, CA 90028

SITUATION WANTED: MANAGEMENT

Have 11 years experience in Pro Audio Manufacturing as General manager, Project engineer, Manufacturing engineer, etc. . . Quality conscious, hard-worker, self starter. Los Angeles area only. Interested companies call (818) 886-4168 for resume.

AUDIO TECHNICIANS

Prestigious national service organization providing recorded educational materials requests resumes from individuals with minimum of two years technical training and experience who can perform electro-mechanical maintenance on recording and duplicating equipment. We are interested in qualified applicants for our future career opportunities. Contact Personnel, Recording for the Blind, 20 Roszel Road, Princeton, NJ 08540.

WANTED FIELD SERVICE ENGINEER

Sony professional audio division is seeking an individual for the position of field service engineer to support the Sony professional digital audio product line. Applicants must have background in audio, digital and video processing techniques. The position is based in Southern California and requires travel. Interested parties should contact (213) 537-4300, extension 436.

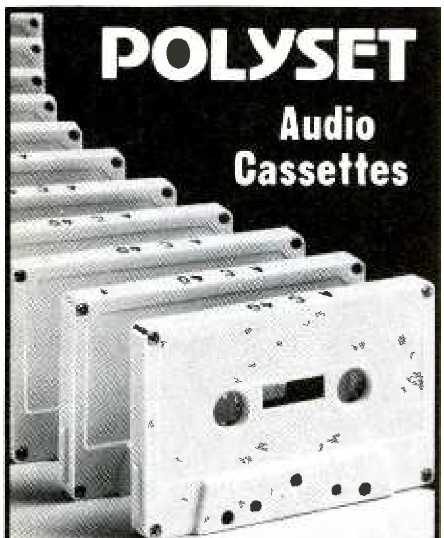
MAINTENANCE ENGINEER

Top Ten Market, PBS Station: Excellent opportunity with growth potential. Min. 3 years maintenance experience, SBE Certification, FCC General Class; College degree preferred; competitive salary.

Resumes to: Gilda Jones
KERA TV/FM
3000 Harry Hines Blvd.
Dallas, Texas 75201

POLYSET

Audio Cassettes



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For additional information circle #155

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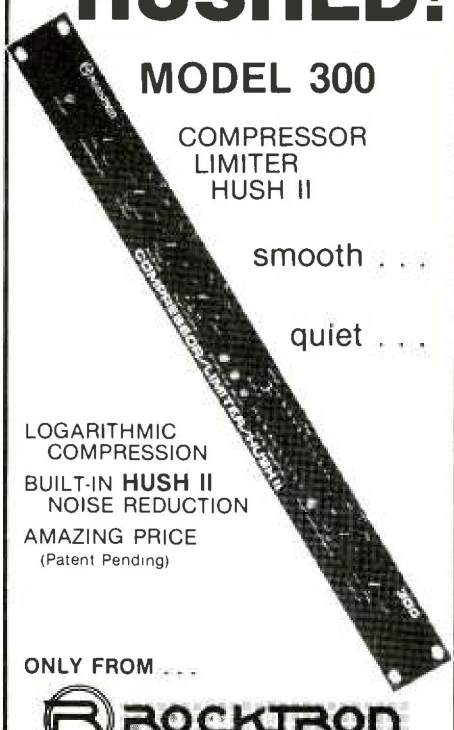

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For additional information circle #156

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Nine out of ten major recording companies choose Electro Sound audio cassette duplicating equipment because it is top-quality and unique. And so is our sales position for this very equipment on the East coast, for the person with the right attributes.

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Send resumes to: David Bowman, Sr. Vice President, Electro Sound Inc., 160 San Gabriel Drive, Sunnyvale, CA 94086. An equal opportunity employer.

Electro Sound
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RECORDING ENGINEER/MANAGER
State of the Art 24 Trk Studio opening August 1 is seeking experienced engineer/manager with impeccable character and reputation to be #1 man. Prefer individual with strong trade following and interest in profit shareing. Interested parties reply to W.C. Wheat, 320 Berkshire Place, Shreveport, LA 71106. (318) 868-8873

EQUIPMENT for SALE

EQUIPMENT FOR SALE
NEOTEK SERIES III: 28×24 with 16 busses, six echo returns, custom "pre-return" submaster fader, extra patch bay, producer's desk, leg set, patch cords, spare parts, exc. cond. Also: White 32-band EQs, $550 ea. (312) 864-4460.

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1081 4 BAND EQ INPUT MOD 1.2 K EA., 2254A LIM/COMP 1.K EA., 1900 SW. MOD. .3K EA., 16 IN. MONITOR 3K. AND MUCH MORE.
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R-e/p Box G
P.O. Box 2449
Hollywood, CA 90078

FOR SALE
HARRISON 4032; Allison Automation; 32x32; 3 Spare IO's; EQ-Shelf+Peak Mod; 7 Sends; DBX 101 VCA's w/Tri-Color LED Read Out. Excel. Cond. Call Tom Paddock, **Different Fur Recording**, 415/864-1967; 3470 19th Street, San Francisco, CA 94110

FOR SALE
Used. MCI 428 Input mods $625 EA. MCI 428 Power supplies $275 EA. MCI 3' Producers desk $375, MCI 428 master & aux mods $525 EA. MCI 428 master & aux mods $525 EA. MCI 428 mainframe w/meters, patchbay $2500, 6 MCI 416 mods $2500. Ecoplate 1 reverb (4x8) $2950, Peavey MC-2, 24 input console w/ road case $1250. **New** AMEK/-TAC consoles, Adams-Smith, Tascam, Milab, ART/MXR, Tannoy. Call!! **CSE Audio, Rochester, New York. 716-227-7763.**

EQUIPMENT FOR SALE
35mm & 16mm Dubbers:
Westrex RA 1551
Solid state with 3 phase sync I/L
Call Bob McDowell at 212-573-6777.

FOR SALE
(1) Ampex MM 1200 24-Track w/Remote and VSO $19,500 (1) Ampex 440-C $2995.00 (1) Syncon Series A 28x24 Console automated w/(2) CPE Compu-Editors $22,500 (2) DBX 216 Noise Reduction at $3995.00 per Unit. 617-685-1832.

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News

— *continued from page 14 . . .*

• **SONY CORPORATION OF AMERICA** recently completed integration of its MCI manufacturing facility at Ft. Lauderdale, Florida, into the company's world-wide manufacturing operations by renaming the facility **Sony Professional Products Company**. Sony acquired MCI, Inc. in February 1982 from its former owner, G.C. "Jeep" Harned, who founded the company in 1955. According to president Henry Klerz, "Sony Professional Products Company will engineer and manufacture a full range of audio and video products and systems." Manufacturing facilities are housed in two buildings with a total floor area of 200,000 square feet.

• **COMMUNITY LIGHT & SOUND** has been acquired by **WHELEN ENGINEERING COMPANY, INC.**, a manufacturer of visual and audible signal devices. CL&S was founded in 1965 by Bruce Howze, and was possibly one of the first companies in the sound-system industry to use fiberglass technology in the production of acoustic horns. Today, it produces a broad range of commercial and pro-audio products from its headquarters in Chester, PA.

• **THE UNIVERSITY OF IOWA** will be holding its annual *Seminar in Audio Recording* from June 24 thru July 5. The program is said to provide intensive training in two, two-hour sessions per day, under the instruction of Professor Stephen F. Temmer. In addition to classroom lectures, there are also demonstration sessions on recording equipment installed at the University's School of Music. Discussion topics will range from fundamentals of acoustics, through microphone theory and standards and measurements, to digital processing and direct-metal disk mastering. For additional information contact Lowell Cross at (319) 353-5976.

• **KEF ELECTRONICS LIMITED,** the U.K.-based manufacturer of monitor loudspeakers, has set up a U.S. subsidiary, **KEF Electronics of America, Inc.** The new operation, located in Chantilly, Virginia, will be headed up by Fred J. Yando, president, and Peter Hoagland, VP/sales.

• **BRUEL & KJAER** has organized a national series of technical seminars and workshops on acoustic measurements, which will be held in major U.S. cities over the next several months. Topics to be covered include basic acoustics and sound measurement; sound intensity theory, measurement, application and instrumentation; plus acoustical noise control. Contact B&K Instruments at (617) 481-7000 for further details and registration information.

• The Academy of Motion Picture Arts and Sciences has awarded the **3M COMPANY** with a Scientific and Engineering Award for the development of Cinetrack 350/351 35mm mag film, a product used in the film-sound recording and post-production industries, and which has the same oxide formulation as Scotch 226 mastering tape. Cinetrak 350/351 is described as offering extended dynamic range, enhanced frequency reponse, lower distortion and improved signal-to-noise ratio, in addition to a superior wear capability. Since its introduction in 1982, Cinetrak 350/351 has been use to record numerous film soundtracks, including *2010, Indiana Jones and the Temple of Doom, Ghostbusters,*and *Romancing the Stone.*

• **RUPERT NEVE, INC.** has published an eight-page booklet, entitled *Buying an Audio Console for Broadcast Video,* which is intended — as the title suggests — to offer a range of practical buyer information, including definitions of common console features; a user's explanation of console automation; and a "real-world" guide to console shopping. The questions addressed by the book range from "Do you really need 24-track monitoring?" to "What grouping and assign functions do you really need?" to "Do you need a console that can be operated in multitrack, stereo or mono configurations?" Also included is advice from several well-known mixing engineers on their own audio-for-video choices. Copies of the booklet are available free of charge from Rupert Neve, Inc., Berkshire Industrial Park, Bethel, CT 06801.

• **QSC AUDIO PRODUCTS** has appointed the following two companies to handle its line of power amps for studio monitoring and live-sound applications: **RJ MARKETING LIMITED,** of Rockville, MA, to cover eastern Pennsylvania, southern New Jersey, Maryland, Virginia, and the District of Columbia; and **NORTH COAST MARKETING,** of Erie, PA, whose territory includes western Pennsylvania, Ohio and West Virginia. Among other lines, RJ Marketing also represents Electro-Voice, Nady Systems and Sequential Circuits, while North Coast Marketing handles Soundcraft, AKG, Fostex and Rane

• **URSA MAJOR** has changed its address to: Box 28 New Town Branch, Boston, MA 02248. Telepone: (617) 924-7697. Telex: 921405.

• **AUDIO+DESIGN/CALREC** has opened a Los Angeles office, which will be headed up by **Dick Swettenham.** Sales and support of Calrec UA8000 Series and Minimixer consoles will be handled directly from the new L.A. office, and Swettenham will offer support to the existing dealer networks for ADC analog and digital processors.

• **AUDIO PRODUCTIVE, INC.** is the new name of **Audiotec, Inc.** According to a company spokesperson, the change has been made to prevent confusion with other similarly named companies in the audio field.

People on the Move
. . . *continues overleaf* —

News

— People on the Move —

• **Andrew A. Brakhan** has been named president and chief executive officer of **Sennheiser Electronic Corporation**. Commenting on his recent appointment, Bracken offered that "Sennheiser has a solidly established reputation for engineering excellence, and a very solid customer base; we will focus our efforts on specific market segments and build upon the existing essential ingredients of factory R&Dknow-how and customer confidencesupport."

• **David Walker** has been appointed to the newly created post of director of operations and development at **Alpha Recording Corporation.** In his new position Walker will oversee the marketing and development of products and product lines for the company's various divisions, including Sonex and Sound-Tex acoustic materials, and The Boss computer-controlled timecode editing system.

• **Mary C. Sauer** has been named as director of marketing at **The Droid Works**, an affiliate of Lucasfilm, Ltd. and Convergence Corporation, with responsibility for formulating marketing strategies for the EditDroid "electronic flatbed" editing system, and the Soundroid audio signal processing system, plus other forthcoming post-production systems from the company. Prior to joining The Droid Works, Sauer was director of marketing of commercial products developed by Lucasfilm's computer division.

• **Gregory A. Green** has been named North American sales manager of the Professional Products Division at **dbx, Inc.** Prior to joining the company, Green spent four years as director of sales and marketing at Ashly Audio, where he was responsible for all domestic and foreign sales.

• **David E. Goldschmidt** has joined the marketing team at **Electro-Voice, Inc.**, with special responsibility for broadcast operations, remote recording and satellite communications for the National Public Radio uplink. He is a recent graduate of Michigan State University, and concurrently was also employed as an audio technician at WKAR, an NPR affiliate.

• **Alan G. Hershner** and **Daniel P. Marchetto** have been appointed as product imarketing coordinators at **Shure Brothers, Inc.** The pair will have primary responsibility for establishing and maintaining contact with audio professionals in TV and radio broadcasting, as well as video and film production. Hershner will be cover the eastern markets, while Marchetto's territory is the central and western markets.

• **Hiroshi Sawano**, executive director of **TDK Corporation**'s Magnetic Tape Division, has been named president of **TDK Electronics Corporation**. Sawano, who joined the company 27 years ago, replaces former president Rocky Kawakami, who has returned to Tokyo.

• Following the recent decision by **TOA Electronics** to divide its Commercial Sound Division into two separate departments, **Joseph V. Green** has been named as head of the commercial sound group, and **Gail Martin, Sr.** as head of the engineered sound group. Green, who was previously with Altec-Lansing and Yamaha, will oversee the company's "open-line" (unfranchised) products, while Martin will be responsible for sales and product design of all of TOA's franchised commercial-sound products.

• **Richard Avery** has been named VP/sales manager at **MIDCOM, Inc.**, the Dallas-based rental, sales and consulting company that also operates a mobile 46-track facility. Prior to his new position, Avery was responsible for design and production control at Interface Electronics.

• **Rick Plushner**, former national sales manager for Sony digital products, has been named president of **Audio Intervisual Design**, the Los Angeles-based sales representatives and rental outlet for Sony digital and related audio products. During his tenure with Sony, Plushner was considered instrumental in introducing the PCM-1610 audio processor and PCM-3324 DASH-format digital multitrack to the American pro-audio market. Also, **Rodney Pearson** has been promoted to the position of director of systems sales at AID.

• **Cary Fischer**, former chief engineer at United Western Studios, Hollywood, has been named western regional manager for **Digital Entertainment Corporation**, the U.S. sales and marketing division responsible for **Mitsubishi** pro-audio digital products, including the X-800 32-track and X-80 stereo mastering machine. ■■■

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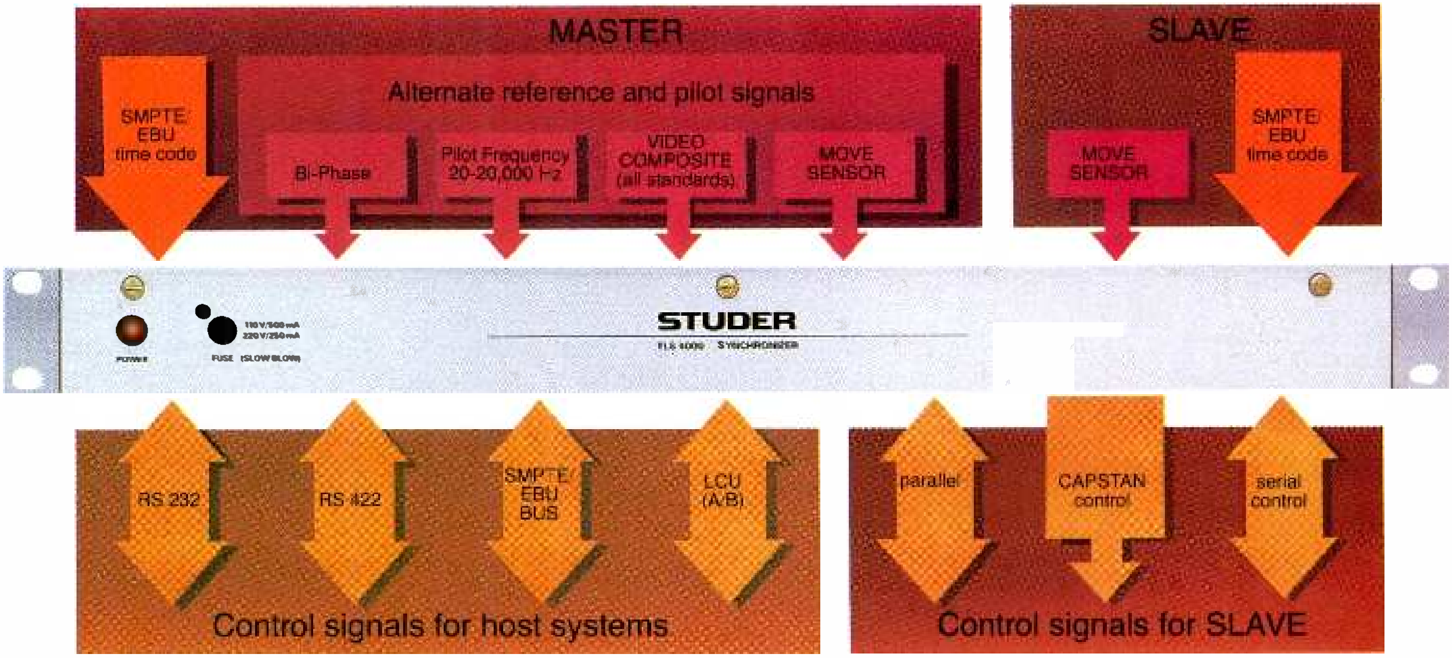
Studer Audio: Production Versatility



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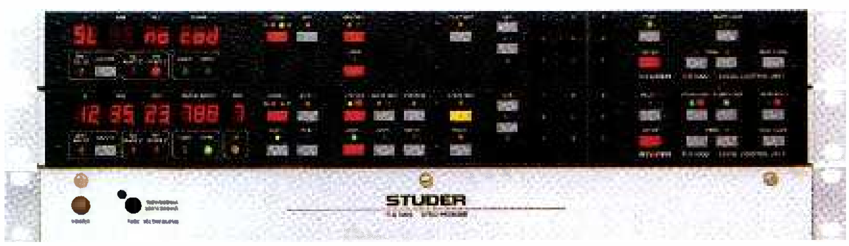

Top to bottom: Type B LCU, Type A LCU, "black box."

For additional information circle #166