November/December 1989 $2.95

db magazine

serving: recording, broadcast and sound contracting fields

Scoring for the Movies

Sigma Sound Studios—*New York*

Lab Report: Fostex D-20

Hands On: *Aphex Aural Exciter III*

& our great gang: Barilla-Hoffner-Learned

Buena Vista at 50

Grand sound... for less than 1/3 of a grand!

The impressive new AT4031 Studio Cardioid Capacitor Microphone.

Our new AT4031 makes a great first impression. And it builds from there. The sound is effortless and natural over the entire spectrum of voices and musical instruments. It's the result of wide, flat frequency response, low distortion, and excellent dynamic range.

Very fast transient response

The outstanding performance is the product of uncommon precision inside the fixed-charge element. The diaphragm is only 0.00016" thick (just 4 *microns*), including the ultra-thin vacuum-deposited gold conductive layer. Very fast transient response and wide bandwidth are assured. With its high sensitivity (– 44 dBm) and low distortion, the AT4031 meets the stringent dynamic range demands of today's storage and playback media.

Uniform, predictable performance

You'll find the AT4031 very easy to use. The uniform, predictable cardioid pattern translates into off-axis attenuation, not coloration. An integral second-order high-pass filter lets you roll off the bass at 12 dB per octave, when wind noise or room rumble must be controlled. An effective foam windscreen is also standard.

No-compromise performance

The AT4031 is intended for those professionals who will accept no compromise in performance, yet need a tough, reliable microphone. But, listen for yourself. Based on extensive field tests in studios and on stages around the country, we think you'll agree that what you hear is precisely what you need...the new AT4031 Studio Cardioid Capacitor microphone.

Circle 10 on Reader Service Card

audio-technica®
1221 Commerce Drive, Stow, OH 44224
(216) 686-2600

November/December 1989 VOLUME 23 NO.6

db magazine

serving: recording, broadcast and sound contracting fields

The recording engineer

THE NEW BUENA VISTA SOUND STUDIOS 16
Chip Murray

SIGMA SOUND STUDIOS—NY 23
Toby Cohen

SOUND RECORDING IN INDIA 42
Sanjeev Kumar

LAB REPORT: FOSTEX D-20 DIGITAL (DAT) RECORDER 57
Len Feldman

TIME-CODE FEATURES OF THE FOSTEX D-20 61
Clay Hutchinson and Dan Hetzel

HANDS ON REVIEW: APHEX AURAL EXCITER III 62
John Barilla

The broadcast engineer

BROADCAST AUDIO: MULTI-CHANNEL SOUND AROUND THE WORLD 10
Randy Hoffner

The sound contracting engineer

SOUND REINFORCEMENT IN NORTH AFRICA—PART II 47
Ed Learned

AUDIO FOR THE CHURCH 52
Brent Harshbarger

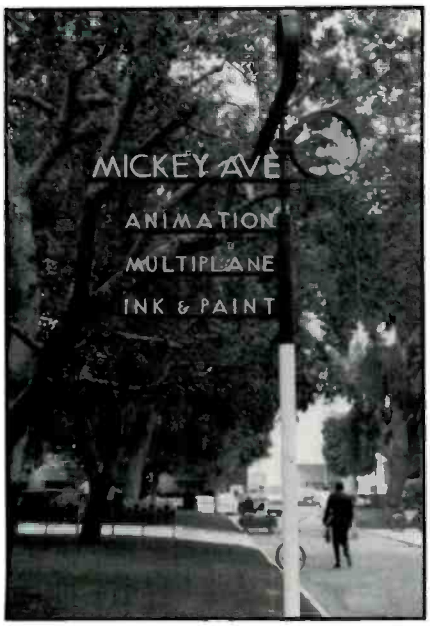
THE ELECTRONIC COTTAGE

HOT TIPS: A MIDI DITTY 6
John Barilla

CREATIVE USE OF THE MACINTOSH FOR AUDIO POST PRODUCTION 28
Frank Serafine

THE CHANGING FACE OF FILM SCORING USING THE "NEW TECHNOLOGY" 31
Arlon Ober

LETTERS 2
EDITORIAL 4
NEW PRODUCTS 66
BUYER'S GUIDE: COMPRESSORS, LIMITERS, EXPANDERS, NOISE GATES, NOISE REDUCTION 71
CLASSIFIED 77
PEOPLE, PLACES, HAPPENINGS 78



About The Cover

● Our cover shows a wide-angle view of the new Stage D dubbing room at Buena Vista Sound Studios of Burbank, California.

How many world-class operations can claim to be located on Dopey Drive? Buena Vista can, since its stages are on that street.

Our article, beginning on page 16 traces how the original Disney stages were transformed into what they are today.

Letters

Department of Corrections, Errors, Mis-Statements, Etc.

• All the typographic gremlins seemed to be active in September/October's cover story *Insights on Mastering*. The transcription of author and Senior Editor John Barilla's interview of Bob Ludwig of Masterdisk got garbled in spots.

TO WIT:

The articles states on page 13 that Ludwig "went on to achieve an M.A. in music literature." In fact, according to Ludwig, "I thought I made it clear (to JB) that I was finishing up work on my M.M. (Master of Music) degree, and had all the credits toward it, but I decided to leave Eastman to go to work with Phil Ramone at A&R before actually getting it. While I won't recommend that decision for everyone, it is one I've never regretted. But I certainly don't want the University of Rochester to think I bestowed a degree upon myself!"

In the paragraph discussing the mastering of *Bruce Springsteen and the E Street Band Live, 1975/85*, Ludwig is quoted as saying "More often than not, Bruce would decide to change some of his phrasing." Ludwig concedes that he may have stated this in the context of a discussion about the album, "...it seems

db magazine

Editor/Publisher
Larry Zide

Associate Publisher
Elaine Zide

Senior Editor
John Barilla

Editorial Assistant
Daniel Buxbaum

Contributing Editors
Bruce Bartlett
Brian Battles
Drew Daniels
Len Feldman
Robyn Gately
Randy Hoffner

Graphics & Layout
Karen Cohn

BPA Audit applied for May 1989

db, The Sound Engineering Magazine (ISSN 0011-7145) is published Bi-monthly by Sagamore Publishing Company Inc. Entire contents copyright 1989 by Sagamore Publishing Company Inc., 203 Commack Road, Suite 1010, Commack, NY 11725. Telephone: (516)586-6530. db Magazine is published for individuals and firms in professional audio recording, broadcast audio-visual, sound reinforcement-contracting, consultants, video recording, film sound, etc. Application for subscription should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year($28.00 per year outside U.S. Possessions, $16.00 per year in Canada)and payable in U.S funds. Single copies are $2.95 each. Editorial, Publishing, and Sales offices are at 203 Commack Road, Suite 1010, Commack NY 11725. Second Class postage paid at Commack, NY 11725 and an additional mailing office. **Postmaster: Form 3579 should be sent to db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.**

to indicate that Bruce, in a live album, would go back and re-sing (re-phrase) a track. What I intended, was that Bruce would go back and use a different performance or change the talking between songs from one performance to another. The article makes it seem that Bruce did not sing *live* what he did sing live. To my knowledge, everything Bruce sang on this famous album was live."

"The article further seems to say that Bruce decided to make these changes based on my input and this is also not true. There are many examples where artists do make changes based on what a mastering engineer's opinion might be, but this was *not* one of them."

STILL MORE

On the bottom of page 14, the article implies that Masterdisk has Neve DTC-1 consoles in most of the rooms. In fact, they have but one Neve DTC-1 console.

At the bottom of page 15, Ludwig is quoted "Only assemble style editing is possible. Insertions are out of the question here...." This is completely incorrect, insertions are indeed possible (in a Sony AE 3000 Editor) and again quoting Ludwig "we do them all the time."

In the second paragraph on page 16 where Ludwig is quoted "You'll need a Harmonium Mundi..." it is of course, a Harmonia Mundi.

Finally, Bob Ludwig has told us "I feel badly that the author left out our other engineers at the studio. It almost makes it seem that I am personally responsible for nearly 25 percent of the Billboard chart. In fact, Howie Weinberg is at least as busy as I am. Scott Hull, Tony Dawsey, Andy Van Dette and Don Grossinger also contribute to making that percentage possible...they do a heck of a lot of work—some for me and a lot they bring in themselves. They and the rest of our staff are a huge part of Masterdisk's success. When they read this article they'll think it's I who is not acknowledging them when, in fact, it was the author."

OTHER ERRORS ELSEWHERE

In the Editorial on page 6, we incorrectly (in the second paragraph) ascribed author Steve Langstaff to an affiliation with the Berklee School of Music. In fact, Mr. Langstaff runs the Audio Workshop School in Cambridge MA, and is not associated with Berklee.

On page 37, in the equipment list for the *Silver Linings* story it is noted that they have an EXR Aural Exiter. While this company no longer exists, it should have been noted that "Aural Exiter" is a registered trademark that only belongs (and only belonged then) to Aphex Systems, Ltd.

Editorial

The large recording studio business is alive and well, thank you. But any observer of the business will know this, but will that observer know just how that business is different today from what it was?

Several articles in this issue point clearly to where the business is today. Witness **Buena Vista Sound Studios** with completely modernized facilities located in the original Walt Disney sound stages. This entire facility, and it is impressively modern world-class, exists for production and post-production of motion pictures.

Sigma-Sound of New York has been completely renovated, including Neve "Flying Faders" and a 32-track digital mastering machine. Why? To do post-production for television, and, to do some major artist music. It's now owned by a major television production house so there will be even more post and sweetening done in midtown Manhattan.

An important and growing aspect of this business is that which has moved from the traditional studio atmosphere to a more personal studio. Recently introduced technology now permits talented artist/engineers to set up and operate sophisticated state-of-the-art equipment at home—in short, an **Electronic Cottage**. The following articles explore that.

First, an article on the march of technology and how it is even changing the business today. **Arlon Ober**, our author, has an impressive list of credits in the motion-picture business as a film composer. Here he examines how much more he can now do on his own home studio equipment, thus bringing a more finished product to the studio.

Frank Serafine, also located in southern California, has graced our pages before with articles exploring his own **Electronic Cottage**. This time he details how the Macintosh computer and a growing software availability, are becoming a focal-point products in the production of music and special effects for television and film.

Let me also draw your attention to our **Lab Report** by Len Feldman (working in his **Electronic Cottage**), beginning on page 57. The review is of the new Fostex D-20 R-DAT recorder with time-code capability. The **Lab Report** is followed immediately by a one-page report by a multi-track studio that used the DAT for master/slave transfers. This is, in turn followed directly by a **Hands-On** report, on the new Aphex Aural Exciter III. It's **Electronic Cottage** studio evaluated by our Senior Editor John Barilla...but we'll let him tell it. LZ

JOHN BARILLA

THE ELECTRONIC COTTAGE

A MIDI DITTY

• Here's a knotty problem for you. Suppose you were working on a major project—an album's worth of material, when the producer in charge decides that all of the snare drum and bass drum parts need to be replaced with better sounds. These parts originally having been programmed on a drum machine could easily be re-recorded—provided, of course, the original drum program is available, but unfortunately it's not.

Your immediate reaction might be, "Well, why don't we just trigger new sounds from the tape and record them on some other tracks." But you find out that you can't do that, because all the tracks are full. (You can't trigger from a track and record onto it at the same time). "Okay", you say, "then let's just trigger the new sounds live as we mix it." That would work alright, except the tape is being shipped out of the country tomorrow to be mixed in another studio, and the producer wants the actual new drum sounds on tape before it leaves the studio.

Does this sound like a tough situation? Well, Tom Schizzano, owner/-operator of *Landford Productions* in East Northport, NY actually faced this dilemma a few months back. But Schizzano "took the (proverbial) bull by the horns". Faced with the possibility of having to call in a programmer to re-program ten songs—with very little time available, he figured out a way to program a drum machine directly from the previously recorded analog tape. It's a nifty procedure that may have some other useful applications.

The following is Tom Schizzano's letter to **db** readers with a few editorial clarifications in parentheses.

THE SOLUTION

"Here's the deal. We took the bass drum and snare off tape and ran them into an MX-1 (a signal conditioner) to make them into clean triggers. Next we took the output of these triggers and put them into a Roland Octapad (MIDI percussion pads) to convert the pulses into MIDI notes.

"Then we grabbed our sync (FSK tape sync) off the multi-track and hooked it up in the usual way to read back tape sync into a DDD-1 (Korg drum machine). At this point, we had sync on one MIDI cable—coming out of the Korg and MIDI note data on another cable—out of the Octapad.

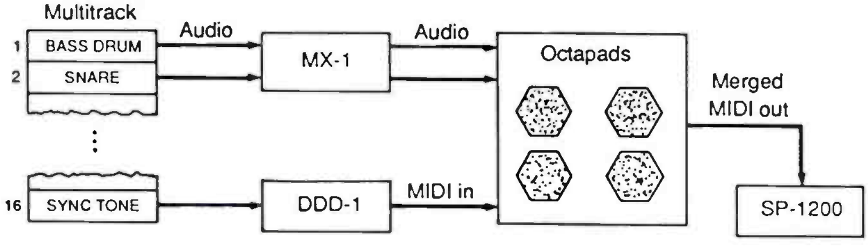
"We merged the note data and sync by plugging the Korg's MIDI OUT to the Octapad's MIDI IN. This gave us the parts and the tempo—both from tape—on the Octapad's MIDI OUT port. This, in turn, was sent to an SP-1200 (E-Mu's sampling drum machine/sequencer).

"The SP-1200 was set to accept external sync clock at high resolution, and was programmed to record the longest bar it was capable of recording. When we ran the tape with the SP-1200 ready to record, sync started and the tune started. (In other words, the SP-1200 was made to go into record in response to the tape and it recorded the MIDI data coming through the Octapads from the original tracks on tape. Got that?) So the multi-track actually reprogrammed these snare and bass drum parts into the SP-1200's memory.

"Finally, we loaded in our new sounds, set the machine to play back this humongous bar (obviously, in response to the FSK sync already on the multi-track), placed the original two (drum) tracks on tape into record and simultaneously replaced the original parts as if someone had spent hours re-programming them.

"After thinking about this I realized there are advantages to this over triggering—even if there had been open tracks. First, there is no trigger delay. These parts go down just like any other sequenced part. Also, if you use a computer or drum machine with sophisticated editing software, parts could be shifted in time, velocity changed etc. (All kinds of things that would not be possible by triggering alone). So the next time you're caught needing to fix drum tracks without the original program, try letting your tape machine program your drum machine!"

Figure 1. This was the best setup.



1990 Editorial Calendar

JAN/FEB

The Professional Electronic Cottage and Broadcast USA—a Synergetic Combination!
Winter NAMM and NAB show issue
GUIDE:
Speakers: performance & monitor

MAR/APR

Sound Reinforcement: theory, and application for various venues—NSCA show issue
GUIDE:
Power Amplifiers

MAY/JUNE

Broadcast, Recording & Sound Reinforcement in Houses of Worship
Summer NAMM issue
GUIDE:
Consoles & Mixers

JULY/AUG

Live Sound—producing it and,or recording it.
GUIDE:
Tape, tape recorders and accessories, Microphones

SEPT/OCT

Audio Post-Production—Television and Film
AES in L.A. Show issue
GUIDE:
Signal Processing Equipment, Part I

NOV/DEC

The Recording Studio—What's happening, what's ahead
GUIDE:
Signal Processing Equipment, Part II, Studio Accessories

BUILDING THE BOMB

The quintessential rock guitar sound. I love it! Years ago, the only way to get it was to crank up a guitar amp to a very unhealthy sound pressure level and stick a mic in front of it. The sound was usually quite electrifying, but unless you had a well isolated studio space, the fallout from your neighbors (or your family) could be equally electrifying.

Fortunately, today we have an array of guitar processors which can simulate the effect of a cranked amplifier very well. It all started with Tom Scholz and his Rockman—the original "heavy guitar sound in a box", and today we have a host of other processors that allow several kinds of distortion (sometimes done with digital algorithms), plus EQ and multiple effects all in single rack mounted unit.

The great convenience of such devices is obvious. We can get a consistently good guitar sound directly on tape—without the hassle of microphones and cables or plugging in a string of devices or even making noise. It's really wonderful the amount of time it saves you.

But for me, (perhaps I'm too much of a purist!) I can't help feeling that something is still missing. The sounds as awesome as they are, still seem a little sterile compared to a mic'ed up stack. What is that missing dimension? I speculate that the major difference is nothing more than air. Many engineers will tell you that air is a very musical transducer of sound. When an electrically-generated sound hits the air it opens up and becomes dimensional. Unfortunately, there is no air between the output of the guitar processor and the input of a recording console.

I decided to test my premise by plugging these devices into a guitar amp and mic'ing them up. When I did an A/B test with the direct output from the processor, there was a noticeable difference. It's hard to quantify this *je ne sais quoi* I associate with speakers/air and microphones, but it renders a realism that I think is impossible to duplicate. I guess what I'm saying is that these processors sound really good, but they sound even better when they hit air before they hit tape.

I also tried this technique with other sounds well known for their blandness, for example synthesized versions of the Hammond organ sound. Even the best synthesized imitations (samples included) are simply insipid. (Ask anybody who's played the real thing.) There's something vitally missing. These patches are just too doggone clean to be real. Putting the synthesized organ sound through an amp and mic'ing it can do a lot to restore the (illusion of) realism. But who wants to have to set up a mic every time you switch a synthesizer patch or plug into your guitar processor? That would be defeating the purpose.

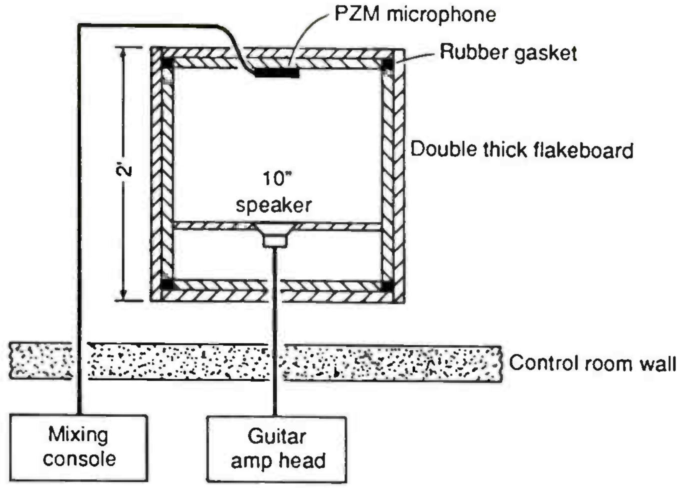


Figure 2. "The Bomb" gives the liveness needed.

So I thought of a nifty way to have that great "speaker- to-air" network constantly on-line without any setup and without generating a lot of SPL. It's simple, and perhaps a little unorthodox, but it works! I call it *The Bomb* because it's such a crude appurtenance to my electronic cottage that no one suspects it could be such an important creative tool.

Here's what I came up with. I wanted to build a "miniature" isolation room for a guitar amp; one that would be always set-up, plugged-in, and ready to record. I also wanted to be able to keep the "guts" of the amp in the control room with me, so that I could tweak the sounds without running around a lot.

So I built a box. A heavy, rigid box about 2 feet cubed. I used ¾-inch flake board because it's dense and it's cheap. To make it extra rigid, I built it double thick—(kind of a box within a box). Towards one end of the box I installed a baffle to mount a speaker. The front end and rear end of the box were made so they could be disassembled with screws for access, but every other joint was both screwed and solidly glued.

So what did I have at this point? An extremely heavy wooden box with a ten-inch speaker mounted on a baffle about a quarter of the way from the back of the box and facing the front. I lined the walls of the unit with glass-fiber insulation and mounted a PZM microphone on the front panel of the box. The removable front and rear panels of the unit were screwed in place with some rubber gasket material sandwiched in between to give it a snug fit. Through small holes in the front and rear, I ran wires for the speaker and the microphone.

This thing was now beginning to remind me of some madman's home-made bomb. So I took *The Bomb* and put it just outside my control room running the wires through the wall. Inside the control room sat an old Fender Princeton amp—(having a nice warm tube sound)—with it's speaker bypassed. I just used the electronics of it to drive *The Bomb*. The PZM microphone installed inside *The Bomb* was then normalled into a channel of my console. (I did try regular microphones, but the PZM sounded best. Apparently, the PZM system of gathering phase-coherent sound really shines in these closed quarters.)

So now anything I plug into the Fender amp in my control room gets shunted to *The Bomb*. The sound is picked up by the PZM and folded back into my console whereupon I listen to it through the control-room monitors. No mics to set up and very little leakage. It's just about as easy to treat as a line-level signal and in many instances sounds a lot more authentic and lifelike. db

DIGITAL PERFECTION

THE MULTIVERB II

A.R.T. proudly introduces a stunning new version of one of the finest sounding effects processors ever made. The Multiverb II offers four studio quality effects at once, pitch transposing, Performance Midi® and new reverb algorithms that border perfection—with noise and distortion figures dramatically lower than comparable units! The Multiverb II offers more power and performance than anything in its class.

FEATURES

- Reverb • Delays • Stereo Panning • Chorus
- Flange • Gates • Reverses • Tap Delays • Eq
- Real-Time Performance Midi®
- Up to four top quality effects at once
- Pitch transposing - 2 full octaves
- The finest sounding algorithms available
- 16 bit state-of-the-art digital to analog converters
- 200 memory locations • Full parameter variability
- Rugged internal power supply
- Midi mapping/Midi in/out/thru
- Ultra wide dynamic range • Full stereo in/out
- Software updateable-product never becomes obsolete
- System exclusive codes-edit/store via computer

THE MULTIVERB EXT

Sampled Perfection! This new super-effector does everything that the Multiverb II does with delay time of two full seconds and studio quality sampling! You can create shimmering layered sounds and sample on the fly using manual or automatic triggering. Besides using four effects at once, you can control everything in real time with Performance Midi®. The EXT provides the amazing reverbs and musical accuracy that A.R.T. is famous for, with the power to create all you can imagine.

ART

APPLIED RESEARCH & TECHNOLOGY INC.

Circle 16 on Reader Service Card

Export Distributors:
THE NETHERLANDS/Audioscript B.V./Soest/02155-2044/FINLAND/MS-Audiotron/Helsinki/90-5664644/SWEDEN/Muskantor & Co./Molndal/031-878080/FRANCE/High Fidelity Services SA/Paris/(1) 42.85.00.40/CANADA/Yorkville Sound LTD./Scarborough/416-751-8481/ITALY/FRATELLI CRASIO/PAVIA/0385 48 029/WEST GERMANY/PME Audio/BAD FRIEDRICHSHALL/07136-6029/ SWITZERLAND/Musikengros P. Gutter/Sissach/061-983757/HONG KONG/Audio Consultants LTD./Kowloon/3-7125251/JAPAN/Nihon Electro Harmonix/Tokyo (03)232-7601/THAILAND/Beh Ngiep Seng Musical Instruments/Bangkok/222-5281/INDONESIA/PT Kirana Yudha Teknik/Jakarta/3806222/SINGAPORE/Lingtec PTE. LTD./7471951/SPAIN/Aplicaciones Electronicas Quasar SA/Madrid/ 6861300/TURKEY/Kadri Cadmak/Istanbul/(1) 1661672/DENMARK/M.I. Musik/Kobenhavn/1-854800/U.K./Harmon U.K./Slough/0753769I1/NEW ZEALAND/Maser Digitek/Aukland/444-3583/4/ ISRAEL/JAZZ ROCK/Tel Aviv/03-24-9941/ARGENTINA/Lagos Sarmiento/Buenos Aires/40-6455/NORWAY/AUDIOTRON/OSLO/2 35 20 96/TAIWAN/Sea Power Co./TAIPEI/2314 3113/ GREECE/ZOZEF/PIRAEUS/01417C151

APPLIED RESEARCH & TECHNOLOGY INC., 215 Tremont Street, Rochester, New York 14608, (716) 436-2720, TELEX: 4949793 ARTROC, FAX (716) 436-3942

RANDY HOFFNER

Broadcast Audio

Multi-Channel Television Sound Around The World

• The United States is not the only country in the world to employ multi-channel television sound broadcasting, nor is it even the first. It is however, true that after a little more than four years of MTS broadcasting, the United States is indisputably the quantity leader in stereophonic television broadcasting. At this point in time, well over 500 of the approximately 1200 United States television stations are equipped for stereo transmission, and virtually every household in the entire U.S. market is within reach of at least one stereo signal. It is reliably estimated that well over 25 percent of the 90 million television households in America have television stereo reception capability. The U.S. Federal Communications Commission authorized the broadcast of multi-channel television sound in 1984.

THE EIA-J SYSTEM

Three countries preceded the United States in such authorization, using two completely different systems. In 1978, Japan commenced multi-channel television sound broadcasting with the **EIA-J** system, an adaptation of which was one of the three proponent systems considered in the United States. In 1981, German broadcasters began using their own multi-channel sys-

tem, which was also adopted by Australia in 1984. In 1986, the British Broadcasting Corporation (BBC) began experimental broadcasting with their own proprietary television stereo transmission system, one which employs digital audio. Let's look at these three other multi-channel television sound systems in use and compare them with the **BTSC** system used in the United States and Canada.

THE AMERICAN SYSTEM

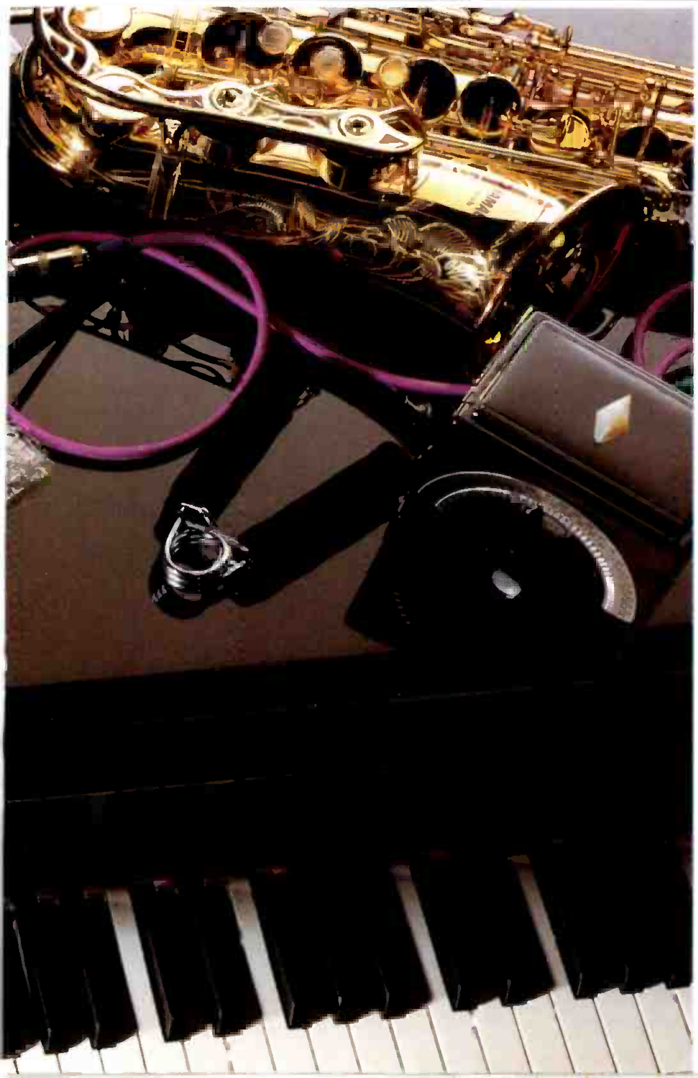
First, however, let us briefly review the **BTSC** system. *Figure 1* is a representation of the **BTSC** aural baseband. Main channel modulation consists of the sum of the left (L) and right (R) stereo channels and deviates the frequency-modulated aural carrier ±25 kHz peak; a stereo pilot at the horizontal scan frequency (f_H = 15.734 kHz) deviates the aural carrier 5 kHz. The stereo difference signal (L-R) is compressed according to the **BTSC** noise-reduction compression algorithm and amplitude-modulates a double-sideband suppressed-carrier subchannel centered at $2f_H$(31.47 kHz), which deviates the main aural carrier ±50 kHz peak.

The second audio program signal frequency-modulates a subcarrier centered at $5f_H$(78.67 kHz) which deviates the main aural carrier ±5 kHz peak. The professional channel frequency-modulates a subcarrier centered at $6.5f_H$(102.27 kHz) and this subcarrier deviates the main aural carrier ±3 kHz. The stereo pilot, the L-R subcarrier, and the SAP subcarrier are locked to the horizontal sync pulses of transmitted video. The **BTSC** system is the only system in use that incorporates noise-reduction companding.

THE JAPANESE SYSTEM

Of the world's other television stereo systems, the Japanese **EIA-J** system bears the strongest resemblance to the **BTSC** system. *Figure 2* is a representation of the **EIA-J** system's aural baseband. The main channel is identical to the U.S. main channel in all respects. This baseband contains only one program subcarrier, centered at $2f_H$ and locked to horizontal sync, as in the **BTSC** system. This subcarrier is frequency-modulated (maximum deviation ±10 kHz) with either L-R signal or a secondary audio program such as a second language. The deviation of the main aural carrier by the subcarrier is ±20 kHz for stereo and ±15 kHz for the dual-language mode. A control signal subcarrier is transmitted at 3.5fH or 55.07kHz. This control signal deviates the aural carrier ±2 kHz and is amplitude-modulated with a 982.5 Hz tone to signal stereo transmission, or with a 922.5 Hz tone to signal dual-language transmission.

In the Japanese system, the control signal's functions are to light the appropriate indicators and to switch the receiver into the proper reception mode; in the **BTSC** system in addition to performing these functions, the stereo pilot furnishes the reference frequency for reconstruction of the suppressed 2fH carrier in the receiver. The Japanese system will at one time accommodate only stereo **or** second audio program, but



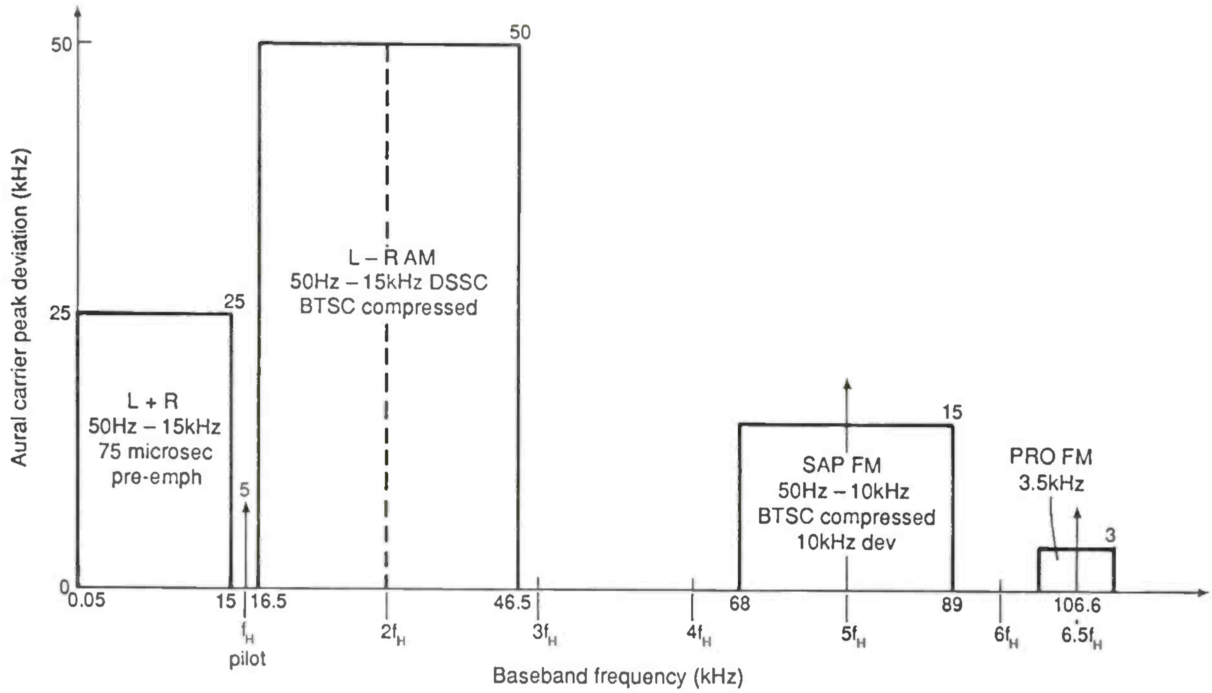


Figure 1. The American BTSC stereo system.

not both simultaneously because the same subcarrier must serve for both services. While the U.S. and Japanese multi-channel television sound systems both add audio channels to the television signal by extending the spectrum occupied by the aural baseband and modulating it with subcarriers, the other two systems actually

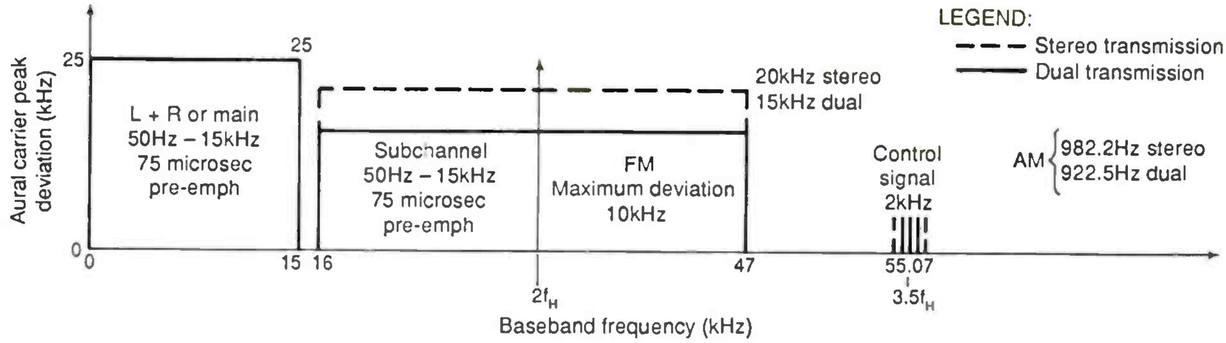


Figure 2. The Japanese EIA-J stereo system. The dual tranmission system is used mostly for Japanese/English tranmissions that are beamed to guests in Japanese hotels.

add a second aural carrier to the television signal.

EUROPEAN SYSTEMS

The German television stereo transmission system adds a second frequency-modulated aural carrier centered 242 kHz above the main aural carrier, and with a power level 20 dB below the visual peak and 7 dB lower than the main aural carrier. The main carrier is modulated with L+R as in the American and Japanese systems. The second aural carrier is modulated simply with right channel audio. L+R and R may be decoded into L and R by the receiver in much the same manner as L+R and L-R. The rationale for using right channel audio rather than a different signal to modulate the second aural carrier is that this produces a higher average level of modulation of that carrier than modulating it with L-R would, resulting in a higher signal-to-noise ratio in that channel.

In the summer of 1986, the British Broadcasting Corporation began experimental television stereo broadcasting using a BBC-developed stereo system that differs appreciably from any of the others. Like the German system, it employs a second aural carrier. This carrier is located 6.552 MHz above the visual carrier, and its power level is 20 dB below peak visual power. The BBC system maintains a discrete monophonic audio signal on the main aural carrier, while the second aural carrier is modulated with discrete left and right channel signals which are digitized, compressed to reduce occupied bandwidth, and transmitted in the digital domain, modulating their carrier using differential quasi-phase shift keying (DQPSK). The digital carrier is actually located slightly above the upper television channel boundary, and is reduced in power level to avoid interference to the upper adjacent channel.

...in addition to the BTSC system of multi-channel television sound employed in the United States and Canada, there are other rather different systems in use in other parts of the world.

The digital audio signal is called **NICAM-3**, an acronym for *Near-Instantaneous Compression And Multiplexing*. The initial resolution is 14 bits, with digital compression to 10 bits. The system's sample rate is 32 kHz, providing 15 kHz audio frequency response. The overall data transmission rate is 728 kilobits per second, and the signal requires about 750 kHz of bandwidth for transmission.

The BBC continues to transmit stereo using the **NICAM 728** system on an irregular basis. Stereo transmissions have only been sent from the BBC transmitter at Crystal Palace, which serves the London area.

The BBC has made no firm commitment to a starting date for regular stereo service, or the expansion of transmission capability beyond the London area; this announcement is eagerly awaited.

The BBC does, however, produce a number of programs in stereo on an *ad hoc* basis, and if a program is available in stereo, the Crystal Palace transmitter transmits it in stereo. The BBC **NICAM 728** system has been adopted as the standard system for television stereo transmission in the United Kingdom, and as a standard system by the European Broadcasting Union. Television receivers containing the **NICAM 728** decoding circuitry are now on sale in England, and television services in England other than the BBC are equipping themselves to transmit using the system.

SUMMARY

Accordingly, in addition to the **BTSC** system of multi-channel television sound employed in the United States and Canada, there are other rather different systems in use in other parts of the world. While the **BTSC** and the **EIA-J** systems have their roots in the subcarrier-based multi-channel sound system first devised for FM stereo transmission in the United States—augmented with some important improvements and adaptations for television; other systems use other means to achieve the goal.

Each system has its advantages and drawbacks, and each in many ways is particularly suited to the television transmission system that it is used with. db

CHIP MURRAY

The New Buena Vista Sound Studios

Before Walt Disney built Disneyland, Disney World or Epcot Center, he constructed a state-of-the-art motion picture studio on 44 acres of mountain valley in Burbank, California.

The year was 1938 and the recent heady success of *Snow White and the Seven Dwarfs* had propelled Disney to the status of world leader in the production of animated films.

The new, intricately planned Disney studio was to be a harbinger of things to come. It was designed as a small, self-contained town with its own streets, storm drains, water and natural gas lines, sewer system and fire hydrants. Modern underground utility conduits routed the electrical and telephone cables. Because of the need of clean air and humidity control for the creation of animated films, the lot included a massive central air conditioning plant.

Constructed along Dopey Drive on the tree-shaded, campus-like studio lot were three buildings devoted exclusively to sound production. Stage A was the scoring stage; Stages B & C, in a single structure, were for dialogue and foley and Stage D was the main motion-picture viewing theater. From their first use in January 1940, these stages have remained the headquarters of Disney's sound operations.

Fast forward to the early 1980's. Forty plus years have taken a toll on the stages. Because these were in-house sound facilities, there was never a compelling need to follow the advances of sound technology demanded of Hollywood's highly competitive independent stages. Disney producers had long left the studio to re-record sound at outside facilities. The sound department was operating in the red and by the mid-1980's had dwindled to a staff of less than ten people. The ADR and foley stages were shut down and used for furniture storage and the studio blood bank.

ENTER JACOBUS ROSE

It's 1986 and Disney Chairman Michael Eisner's administration has been in office a year. The company is in the early stages of a major transformation and turns to Rose, then 29, the former Vice President of Sales, Marketing, and Operations at Glen Glenn Sound, for advice on what to do about Disney's aging sound facilities. "When I first saw the facilities I couldn't believe it," recalls Rose. "The whole studio sound department had lived in a bubble for 40 years."

Aggressive, smart, and impatient, Rose signed on with Disney as Sound Director and quickly went into action. He forged a coalition of top industry sound re-recording mixers and their loyal producer clients to approach the studio officials with a proposition: If Disney would refurbish and upgrade the sound facilities, the mixers and producers would bring sound post-production of their films and television programs to the new facilities on the Disney lot. The largest of these producers, Stephen J. Cannell, was in a position to deliver a sizable chunk of prime-time television programming to such a facility.

Based on that promise, Disney officials funded the renovation of the studio's original sound facilities and began to transform it into an independent facilities company called Buena Vista Sound Studios. The new entity would service Disney productions as well as outside clients.

Now, after nearly four years, all of Disney's original stages have been refurbished under the multi-million dollar restoration program. They represent the most modern and technically-advanced dubbing, foley, and ADR stages in the motion-picture and television industry, and have already catapulted Buena Vista Sound into the "big three" of Hollywood post-production sound companies.

RECENT PRODUCTIONS

Both Disney and non-Disney feature films have recently completed sound post-production work at Buena Vista Sound. *Rain Man* from United Artists, *Leviathan* from MGM/UA and *The Seventh Sign* from Tri-Star Pictures used the newly-renovated facilities. And Touchstone, the Disney-owned motion-picture arm, used the new stages for *Big Business*, *Cocktail*, *The Three Fugitives* and *The Good Mother*. Upcoming features include: *Welcome Home, Roxie Carmichael* and *Blaze*.

On the television side, Buena Vista Sound provides ongoing audio services for *21 Jump Street* and *Wise Guy* for Cannell; *The Magical World of Disney* for Walt Disney Television, and on a project basis, *Dynasty* and *Nightingales* for Aaron Spelling Productions.

Knowing that recruiting top mixing talent draws producers to a facility, Rose has personally supervised a major campaign in recent months to attract the best and the brightest to Buena Vista Sound. As a result of the effort, a major contingent of Hollywood's sound mixing talent has shifted to Disney's lot since the renovation began. From Todd A-O/Glen Glenn Studios came mixers Terry Porter, Mel Metcalfe, Dave Hudson and later Rick Ash and

Dean Zupancic. John Reitz moved from Lion's Gate Studios and ADR/foley mixers Doc Kane and Tim Hoggatt came from Warner Hollywood.

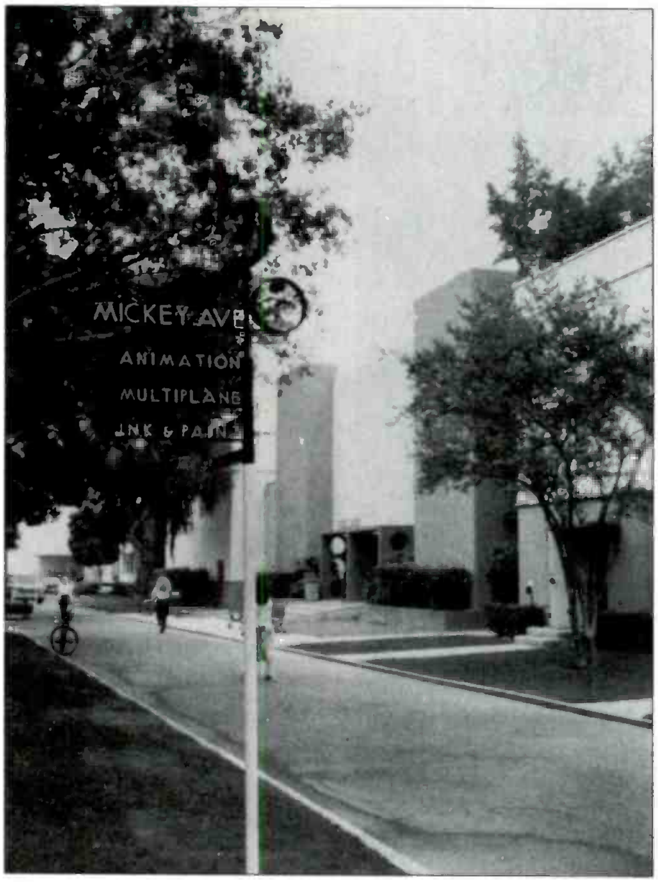


Figure 1. Looking down Dopey Drive. Along this street, the Stages run from front to rear A, D, B, and C.

STAGE A

The first building to undergo renovation was Stage A, Disney's original scoring stage where the music was recorded for such Disney films as *Mary Poppins*, *The Absent Minded Professor* and *Splash*. Sonically, the all-wood building was already one of the best scoring stages in the industry, but the renovation would turn it into a sorely needed dubbing stage for film and television.

With the help of the architectural firm of Backen, Arrigoni & Ross and the acoustical consulting firm of Charles Salter Associates, both of San Francisco, the first phase of the restoration began. From an acoustical perspective, the original sonic qualities—a live room for orchestral recording—had to be altered for the new application of re-recording monitoring.

"Orchestra scoring has an entirely different set of acoustical criteria from a re-recording stage," said acoustical designer David Schwind of Salter Associates. "A scoring stage needs a fairly high degree of liveness so the orchestra can hear itself and play in synchronization. The requirement for mixing is entirely different. You want the most accurate reproduction of sound you can get. The intelligibility of dialogue is one of the most important factors."

ACOUSTICAL CHANGES

The new Disney renovation followed the acoustical design criteria of Lucasfilm's THX monitoring system. Reverberation time measurements were made. Test equipment used in examining the reverberation included a Bruel and Kjaer sound level meter and Hewlett-Packard signal dynamics analyzer. Other than the need to reduce the "liveness" of the room, the results showed no severe acoustical problems on the stage.

"The acoustic changes primarily were in the walls," said Schwind. "We had to add a substantial amount of sound absorbing material. You can't just go in and glue acoustic tiles on all the walls and ceilings and hope to arrive at an acceptable result because you need to look at it frequency by frequency from bass through voice range through the highs."

Schwind said he used between seven to ten buildups of various acoustical treatments from a variety of manufacturers. The materials ranged from glass fiber insulation to hardboard. All exterior wall surfaces were covered by acoustically-transparent, aesthetically pleasing stretch cloth to create the look of a solid wall. An IBM personal computer with custom software developed by the Salter firm allowed a detailed sonic analysis of all the materials used in the construction.

WOOD FLOORS

The original wood floors from the old scoring stage were an advantage. "We've found wood floors tend to give some low-frequency absorption but also some low-frequency re-radiation right into the seat chair, which tends to envelop the listener more into the picture," said Schwind.

Renovation of the structure included addition of a producer's office, kitchen and seating for 125 people. "We redesigned the rear wall in a half circle pattern and padded it heavily to avoid secondary reflections back to the mixers," said Rose, who acted as general contractor for the renovation. "The acoustics were designed to maintain the live feel without any extraneous reflections through a combination of hard and soft surfaces and diffusion panels."

Though instrumentation was used to help make the acoustical design decisions, in the end, human ears made the final judgement on the acoustic renovation. "The instruments process things in a way which is physically accurate," said Schwind. "But the human hearing mechanism is so non-linear that there's no instrument yet available which measures perception. Actual listening tests are still very important in assessing the qualities of a room."

NEW EQUIPMENT

Stage A was outfitted with completely new sound equipment that gives the capability of video and 35 mm film-based dubbing and screening for stereo television and feature films. A custom THX monitoring system utilizing Electro-Voice components was installed. A Harrison PP-1 three-mixer audio console created comfortable work areas for the dialogue, music/foley and effects mixers. A Sony high intensity 9 x 12-foot video image projector was included for viewing tape masters on the theater screen. Behind the huge audio console, a mixer can easily reach an arsenal of processing equipment by BBE, Aphex, Eventide Clockworks, DBX, Urei, Fullmost and Lexicon. Just behind Stage A, in the machine room, 33 Magna-Tech Model 10000 and 2000 series 35 mm reproducers and recorders are linked with Otari 24-track analog and 32-track digital tape machines and Sony ¾-inch video players with SMPTE address track time code.

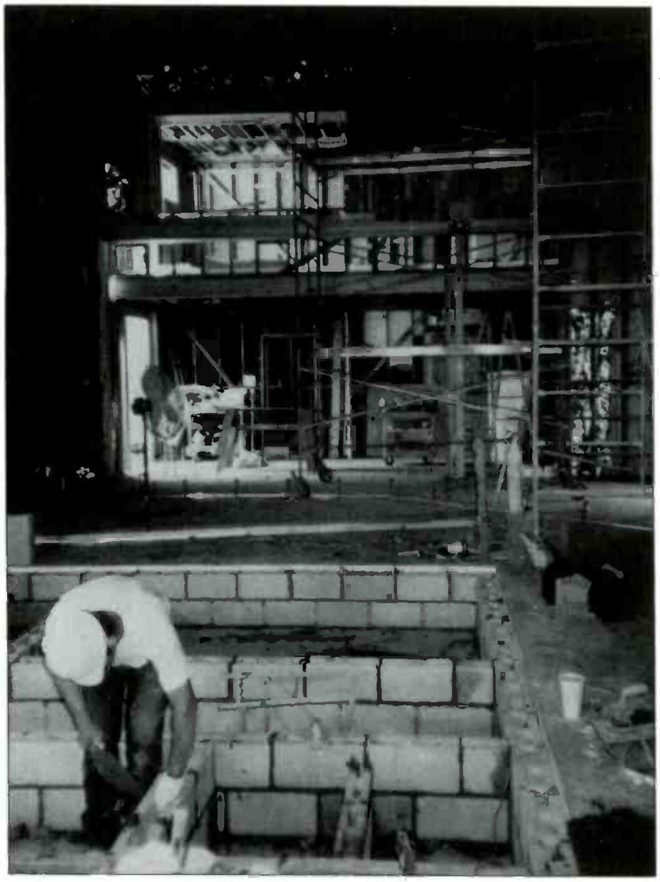


Figure 2. Foley pits for Stage B are shown under construction on an isolated floating floor. The projection booth and control room may be seen in the background.

All machines are controlled by a JSK Engineering MC211 Motion Controller and smart distribution amplifier system which provides motion control for film and tape machines in dubbing or looping. This allows the machine room operators to manipulate any combination of picture or sound elements from tape or film sources.

The JSK Engineering system also has an SDA521B advance/retard feature for use with the Magna-Tech machines which allows the reproducers to be offset while the system is moving or stopped. By being the first stage restored on the Disney lot, Stage A became the *de facto* standard of reference for renovations to come. Rooms must match enough that acoustic judgements made on one stage will not change on another. In order to accomplish this, reverberation measurements made after construction of the first theater were scaled by size to determine the requirements for all subsequent rooms. "This is one of the primary reasons for getting us involved in the first place," said Schwind. "The stages are far from identical but the acoustic characteristics are similar. If the level of dialogue is too low in one room, you'll know it in another. You will never come to opposite conclusions because of room acoustics."

Figure 3. Stage A. Dubbing the film "Gross Anatomy" with actors Mathew Modine and Christine Lahti seen on screen.

STAGE D

Next door to Stage A is Disney's main theater (known as Stage D). It was in this historic room that Walt Disney viewed all his films and had "Fantasound," the first motion-picture surround sound system, installed to properly play the new technology audio tracks on the 1940 animated classic, *Fantasia*. It was also here, in 1967, that Walt Disney watched the Disney Players act out the film script to the movie musical, *The Happiest Millionaire*. The feature was to be the last personally overseen by the filmmaker.

In addition to its historic value to the Disney legacy, the theater was designed by S. Charles Lee, a leading theater designer of the 1930s and 40s. Its architectural aesthetic—*art moderne*—had often been copied and was something project architect Guy Chambers of Backen, Arrigoni & Ross wanted to preserve.

Though the room was a thing of beauty, its acoustics were far from it. "We knew there were some serious acoustical problems and felt we could resolve them and save money in the process," said Chambers. "The challenge was to bring the theater up to contemporary acoustical standards which far exceed those in the day when the theater was originally designed."

Turning the legendary building into Hollywood's finest dubbing theater would be the "crown jewel" of Rose's plan. But after stripping the long, narrow 650-seat theater to its structural spine, Rose knew he was in trouble. "We found when we opened up the walls that the building needed a lot more acoustical treatment than we originally thought," said Rose. "We needed more money and were headed significantly over budget."

At this point, Disney's President, Frank Wells, asked for a list of alternatives as to what could be done with the building other than completing it as a dubbing stage. Rose's vision for Buena Vista Sound seemed near the edge of doom. "The list of alternatives included a simple movie theater, a meeting room or a bowling alley," Rose recalled, wincing at the memory.

Figure 4. Working the JSL Machine Controller.

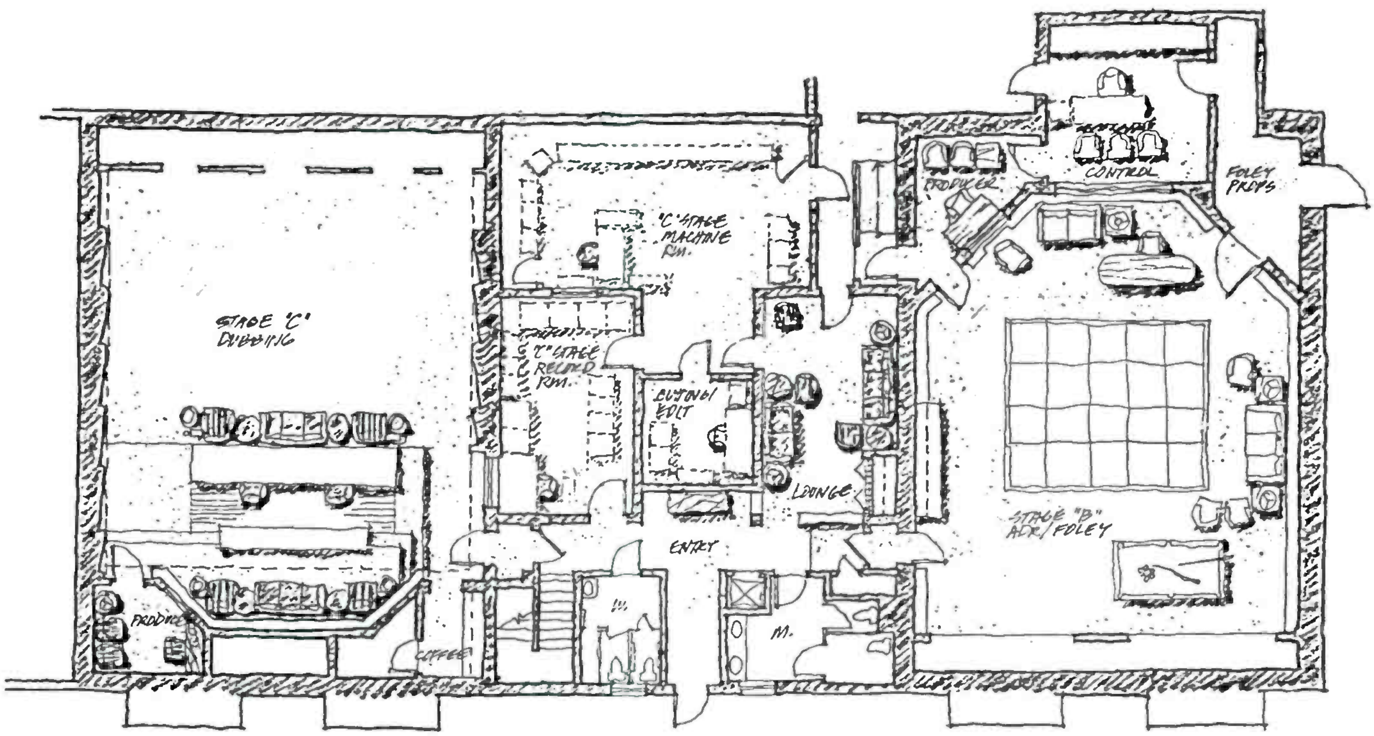
THE DAY OF RECKONING ARRIVES

Frank Wells was to inspect the building and make his decision. Wells arrived, recalled Rose, and found himself standing in some just-poured wet cement. It was at this point that the decision came: "We're not going to make this a bowling alley...just finish it!" Rose, who can laugh about it now, notes that Wells' footprints remain in the cement to this day.

Though built at the same time as Stage A, the main theater differed in that it was constructed mostly of concrete and plaster. An acoustically porous plaster was used on the interior walls as sound treatment. However, after years of re-painting, the plaster had lost its sonic properties. In later years acoustic tiles had been placed on the walls, covering the original plaster. "The acoustic tiles gave a rather low reverberation time at very high frequencies and at low frequencies it was excessively long," said Schwind.

Assessment of speech intelligibility was "very bad," Schwind added. Before restoration of the theater began, the RASTI (Rapid Assessment of Speech Transmission Index) was about 50 percent. The index for a good theater should be about 90 percent, he said. "It was obvious to those trying to mix in the facility that it was not up to the current standard of the industry," said Schwind. "Improvement had to be

Figure 5. The architect's drawing of the Stage B and C complex.



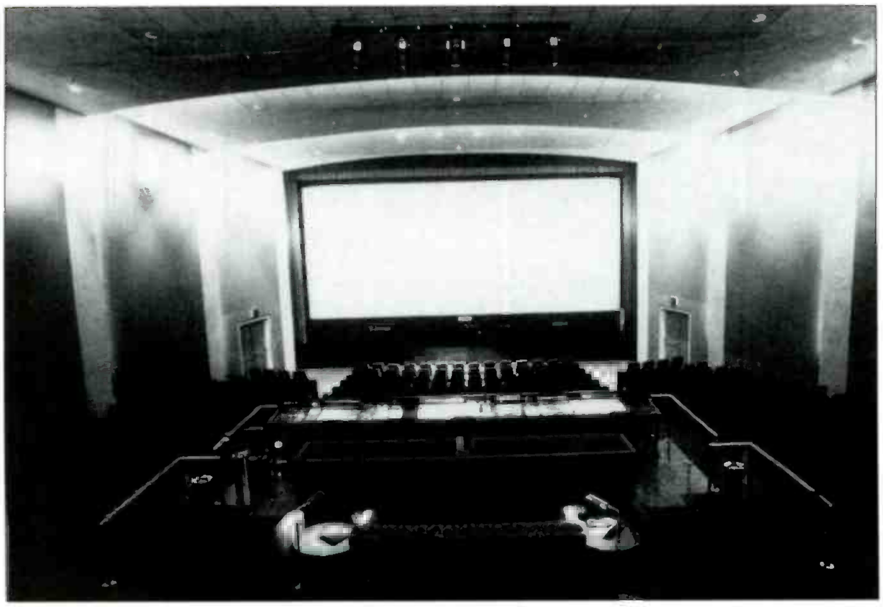
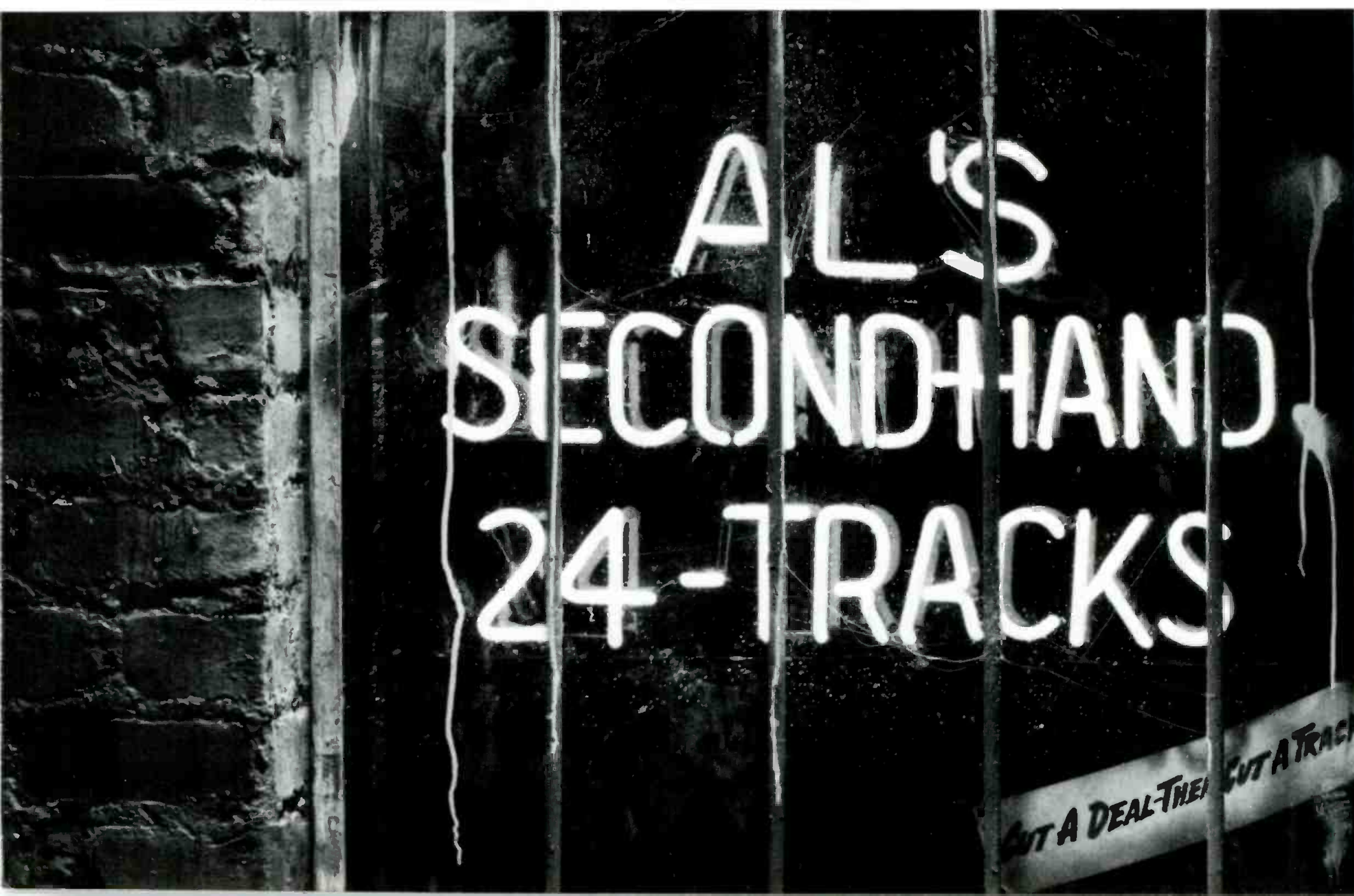


Figure 6. A wide angle shot of stage D.

dramatic and it had to be noticeable to all users. That was very fundamental to our work."

In order to shorten the low-frequency reverberation time, large cavities were cut into the original plaster walls, creating the look during construction of the ribs on a skeleton. Various configurations of sound absorbing treatment was placed in the cavities. Over the openings, acoustically transparent fabric was stretched and shaped to restore the visual appearance of the room. Sections of the original plaster were retained; the result was an interior which retained the original *art moderne* aesthetic but which had vastly reduced reverberation time in the low-frequency range. At 63 Hz, a 2.8 second reverberation time was shortened to 0.87 seconds. At 125 Hz, the original 1.8 second reverb time was cut to 0.70 seconds and at 250 Hz, the time was reduced from one second to 0.63 seconds.

To create acceptable stereo imaging for the mixers, the long, narrow hall was shortened by moving the screen forward about 20 feet. Seating was cut from 650 to 450. A new, curved rear wall was constructed a few feet from the existing rear wall. It was heavily padded to reduce sec-



ondary reflections.

The lobby was restructured, a producers' office and kitchen was carved out of the space, and the machine room was completely renovated. The projection booth was updated with automated 35 mm and 70 mm equipment. The entire building was completely rewired, including cabling connections for five 32-track digital tape machines. Again, a three-position Harrison PP-1 console and custom THX monitor system was installed with the full complement of processing gear used in Studio A. The machine room received 32 Magna-Tech 35 mm reproducers, three 35 mm recorders, an Otari 24-track analog tape machine and three Otari 32-track digital tape machines.


Figure 7. The Stage D machine room.

THE THIRD PHASE

In early 1989, renovation began on the third phase of the project. The original ADR/foley stage was stripped to its foundation and outer wooden frame. Within the footprint of the old building, two new stages, one for re-recording and the other for ADR/foley, were constructed. Essentially, these stages were brand new; all new drywall and absorbing treatments were installed. Again, the THX acoustical criteria was followed and each of the new rooms had the similar reverberation characteristics of the original stages. The

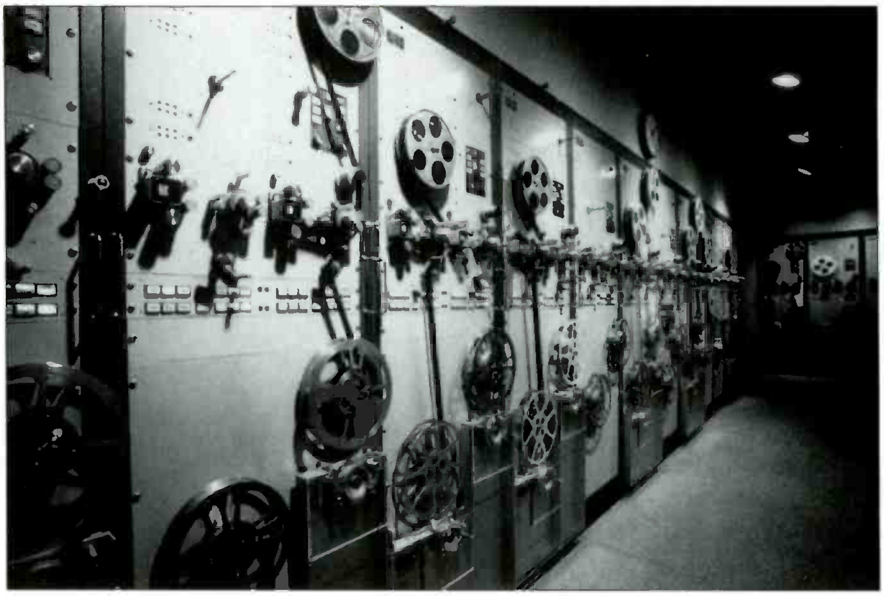


Figure 8. A closer view of the Stage D machine room's Magna-Tech film equipment.

new facility has a central lounge between the two stages and each has private producers' offices.

On the building's top floor, nine film or tape-cutting rooms were built, each wired to each other and to the central machine rooms. In anticipation of digital hard-disk technology becoming more prevalent in film production in the coming years, the rooms were pre-wired for easy switch-over to digital systems.

Though the new stages did not have the architectural preservation problems presented by the theater renovation, Schwind said major constraints came from the original footprint and outline of the existing wooden frame.

"This may not sound like much of a restriction, but it becomes a restriction when you are stuck with how much load can be carried on the existing frame, how much space you have and how to get certain sized duct-work and mechanical equipment in the building," he said.

"It influences the background noise level."

STAGES B AND C

Stage C, the new facility for dubbing, was outfitted, like the earlier stages, with video or 35 mm film-based capabilities for stereo television and feature film. A 5000 series console from Solid State Logic is the centerpiece for the new stage. Monitoring is done through a custom THX system. The machine room houses 33 Magna-Tech reproducers and recorders which are locked to Otari MTR-90 and MTR-12 audio tape recorders. Dolby SR and A-type noise reduction is available. Sony video projection and tape playback is also available.

Next door, Stage B is the video or 35 mm film-based ADR and foley stage for stereo television and feature films. This room features a custom-built Larson Technology/Sound Workshop ADR console with software which allows a wide range of dialogue and looping work. A JSK Engineering motor control system allows programmable manipulation of 35 mm film, 24-track, 4-track and 2-track audio formats in sync to either 35 mm film or videotape.

In addition to the new sound facilities, Disney has put many of its other studio facilities under the umbrella company Buena Vista Studios and is aggressively marketing production services to outside producers. Buena Vista Studios now offers five production stages, visual effects, lighting, grip, props, set construction, paint shop, transportation and wardrobe. Also offered are the studio's backlot and location shooting at the Golden Oak Ranch in Newhall, California.

PARTIAL EQUIPMENT LIST:

Harrison PP-1 mixing console
Solid State Logic 5000 mixing console
Larson Technology mixing console
Lucasfilm Ltd. THX monitoring system with Electro-Voice components
Magna-Tech model 2000 and 10000 series film recorders/reproducers
Otari MTR series 24, four and two-track recorders
JSK Engineering MC211 motion controller with advance/retard
Dolby SR and A-type noise reduction
Sony VO-5850 video recorder/players
Sony Super Bright Video Projectors
Echo, reverb, delay and processing systems by BBE, Aphex, Eventide Clockworks, dbx, UREI, Fullmost and Lexicon

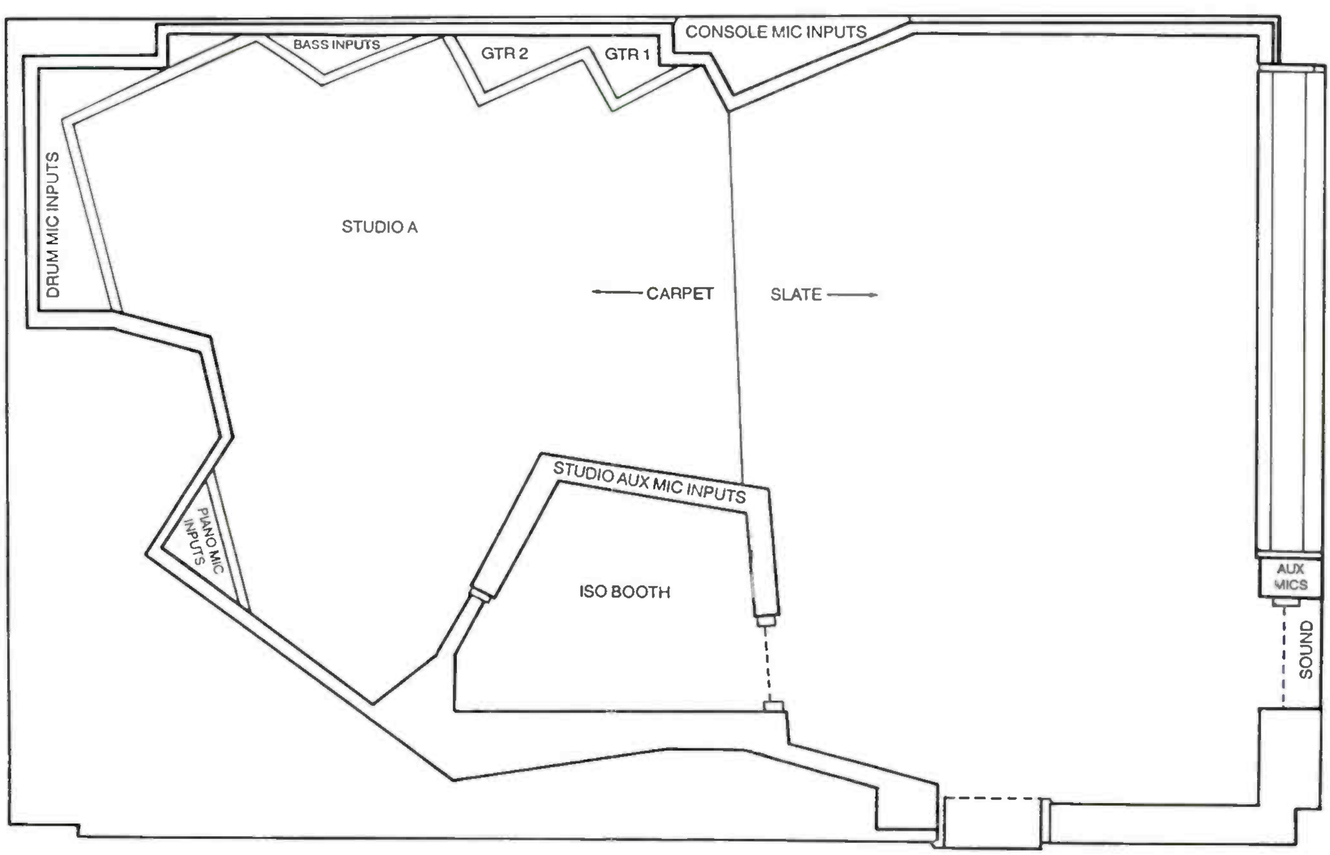
TOBY COHEN

Sigma Sound Studios—NY

"It has a great reputation and a wonderful history—a very significant history," remarked Gary Robbins, general manager at Sigma Sound Studios of N.Y.

Robbins, who came to Sigma upon the recent acquisition of the landmark by M&M Syndications, Inc. (a N.J.-based broadcast product distribution company that also owns N.J. based Edit Masters), reflected that it wasn't in the business plan to acquire a recording facility. "It was more a combination of elements," he said, "whereby the opportunity presented itself. Under those conditions, we looked at the situation and saw it as a good opportunity and something that could be advantageous."He explained the circumstances behind the takeover. "A bit over three months ago, I had received a flyer in the mail from Sigma Sound Studios, N.Y. that said (words to the effect) that Sigma was changing direction and having an equipment sale. It also said to contact Bill Sisca for information. I knew Bill

Figure 1. Floor plan of Studio A.



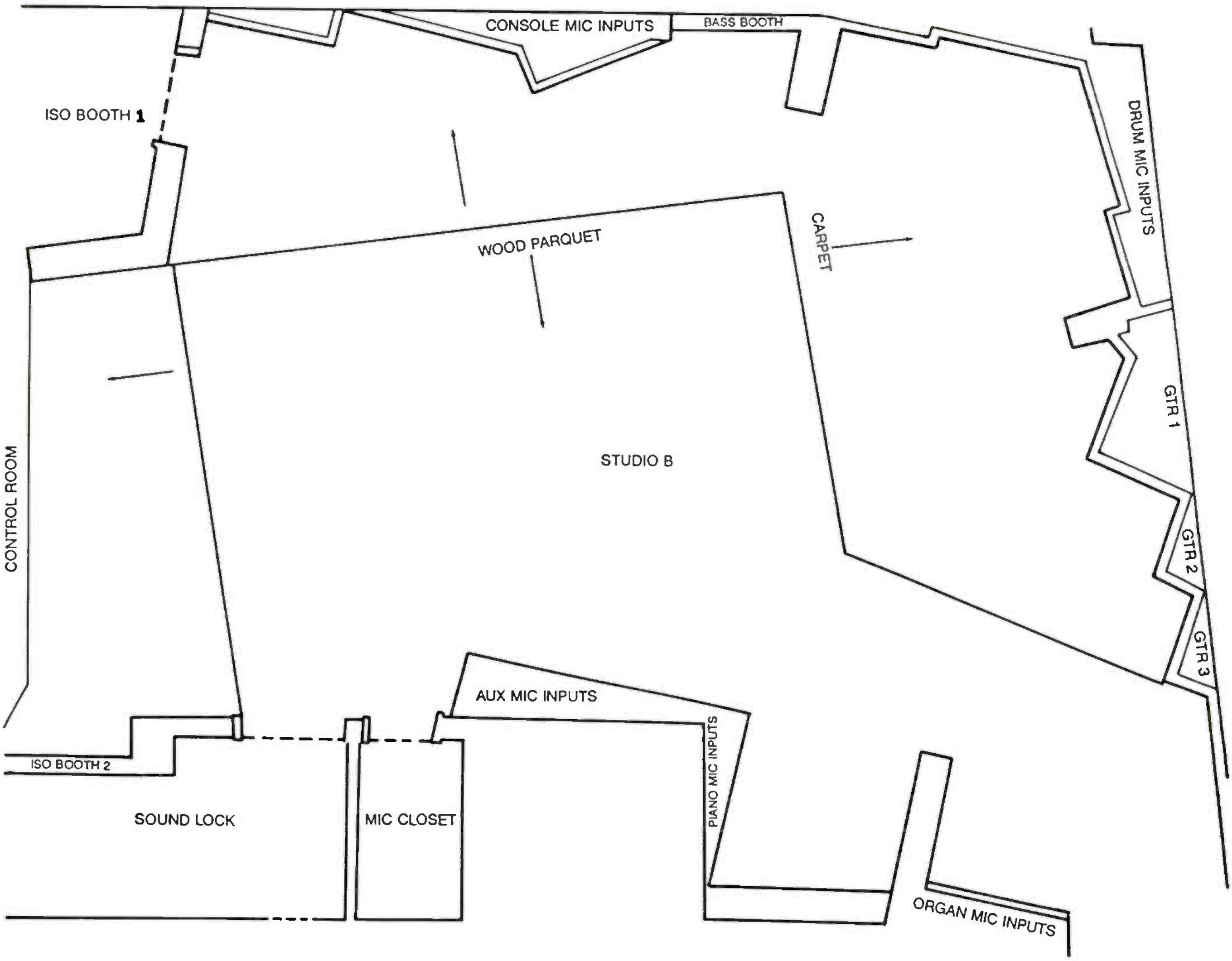


Figure 2. Floor plan of Studio B.

Sisca from previous work. Bill had worked as an independent video producer at Edit Masters, which is our post-production house in the N.J./Philadelphia market. I gave him a call and expressed interest in purchasing some microphones for the voice-over booth at Edit Masters."

With the sale slated to take place the following week, Sisca sent Robbins a FAX of equipment that was approximately twenty-five pages long, as well as an invitation for him and M&M President Michelle Pruyn to take a look at the equipment. They viewed the facility at 2:00 pm on a Thursday, were very impressed, and stayed until about 8:30 that evening.

"We met with Joe Tarsia, who was one of the owners," said Robbins, "and by 8:30, we had decided to buy the place. The rest is history."

SIGMA SOUND'S RECENT PAST

Never having been a specialty studio, Sigma Sound, for about the past ten years, has been servicing a variety of markets, with an extensive base in the record, commercial, film, and tele-production industries. Sigma Staff members have always prided themselves on their ability to service any client's needs, from any background, for any purpose. "We expect," projected Robbins, "to have a very strong music base, and to continue to do commercials, as well as broaden the client base with the video work that we can do." He emphasized that the capabilities of Sigma, M&M, and Edit Masters will all complement each other.

Formed approximately six years ago, M&M Syndications, Inc. produces various television shows and syndicates them to stations across the country. One such show is *The Record Guide,* which, according to Robbins, is hosted by different music personalities every day, and also involves some music video. "After having posted the show for fifty-three weeks," he reflected, "we'd determined that we, in fact, could have bought the equipment to post the show, as well as had our own in-house facility."

And so Edit Masters was formed, based upon the idea that it would be a simple in-house facility, and that special effects and elaborate post work would be done at another facility.

"We went to NAB and spoke with the people at Ampex, and they were very helpful in providing us with a complete system. We came away with a much more complete facility than we had anticipated," said Robbins.

"Edit Masters," he added, "is a production and post house. The facil-

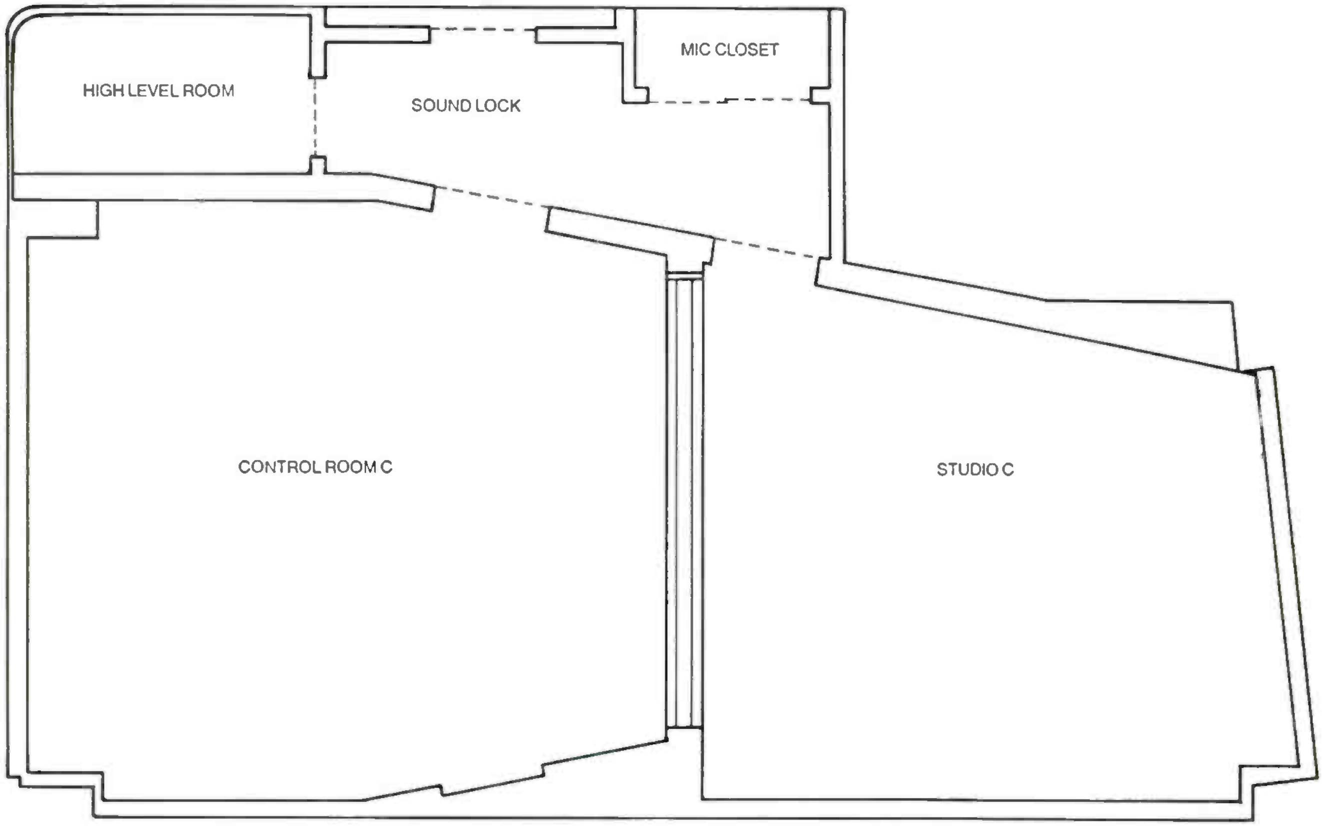


Figure 3. Floor plan of Studio C.

ity produces the television programming that M&M distributes, as well as work for outside clients—everything from corporate and industrial video to movie-production for NBC or CBS."

Figure 4. Studio B control. The Neve VR60 with Flying Faders are seen with a Mitsubishi 32-track digital recorder.

SYNERGISM

Robbins pointed out that Sigma has already done an elaborate thirty-seven hour audio-for-video sweetening job for a major Edit Masters client. He went on to say that Edit Masters is "primarily video-based, and has not, to date, had a need for elaborate audio. With the acquisition of Sigma, however, it made very good sense to have that available to our clients. It can only enhance the services that we provide to them, as well as any internal work that we do.

Edit Masters operates twenty-four hours a day, six-and-a-half days a week. The first shift is for outside clients, the second is for M&M work, and the third shift is for dubbing and maintenance." He continued. "I get back to the idea of the three companies complementing each other. We have the ability now for a client—perhaps a music artist let's say—to not only record an album, but to also promote the music via television programming, or even create a special for them, as well as a music video. So, it's a full complement of services, and it's all interrelated."

One upcoming joint project will be an M&M-distributed show called Studio Sounds, which will emanate from Sigma Sound Studios of N.Y. "The premise," said Robbins, " is to present a show that takes place in the environment of the studio (which is known to people in the industry, but is really sort of foreign territory for music lovers). The idea is to create that kind of excitement and special intimacy that exists in the studio where music is created." The show will feature in-depth interviews with various music personalities and take place in the studio control room. There will also be live performances which will be taped in the studios.

When Sigma Sound Studios first took over the space occupied by Broadway Studios at 1697 Broadway (the Ed Sullivan Theater), it was re-configured and reconstructed to suit the state-of-the-art in technology at that time. Originally, the studio had been equipped with MCI consoles and 3M tape machines, all of which have since been replaced.

STUDIO B

Studio B, roughly 1,200 square feet, is used as a 24-track recording room, as well as a mixing room. It's equipped with an automated 40-input Solid State Logic 6000 console with total-recall computer. There is a 2-inch Studer A-800 Mark III 24-track machine, as well as a ½-inch stereo Studer A-820 mix machine. Studio B also contains an Ampex ATR-102 ¼-inch 2-track machine and an Ampex ATR-104 ½-inch 4-track machine.

Also situated in the room are Pultec and API equalizers, a UREI LA-2 compressor/limiter, a Drawmer 1960 compressor/limiter, Lynx Time Line synchronizers, as well as a JVC 8250 ¾-inch video machine and a color monitor. Studio B has the capacity to handle up to thirty orchestral pieces, and is used for tracking and mixing for tele-production, commercials, film mixes, and video post work.

STUDIO A

Studio A, between 700 and 800 square feet, is the same as Studio B in terms of outboard equipment complement, as well as multi-track and mix machine complement. It also has production capabilities similar to Studio B.

Both studios B and A share a Mitsubishi X-850 32-track digital mix machine. Studio A housed a Neve V-III (with Necam 96 automation), and has now been replaced with the Neve VR 60-input console (with Flying Faders automation). "We will be the first facility in N.Y.C.," explains Robbins, "to offer this very important board. It's essentially the V-series console with a recall feature that's very similar to the one that's available on an SSL board. By pushing a button you can record the position of every single knob, button, and fader on the console, and have the video display of that positioning. That way when you stop a session, you can record (in snapshot form) the console setup, go back to the console at any time, play the disc back, and recall the settings so that they can be set precisely as you had left the board."

Robbins calls the Neve VR a "tremendous time-saver and a very important artistic tool as well, because now the engineer, the producer, and the artist can get back to the mix." He added that the "Neve has always been recognized as a very musical-sounding console. Now, with the recall function, it's the best of all worlds."

STUDIO C

Studio C, primarily a mix room with overdubbing capabilities, is about 300 square feet. It has an automated Solid State Logic 6000E mixing console with a G-series computer and total recall. The console has 48 standard input modules with a bank of 8 stereo modules for a total of 56 inputs. As well, there is a Studer A-800 multi-track.

Also in Studio C is an Otari MTR-90 Type II for 48-track analog work, as well as a Mitsubishi X-850 32-track digital machine. Floating equipment (used on a session-by-session basis)includes two Mitsubishi stereo mix machines and a pool of outboard equipment. Each of the rooms also has access to EMT 140 and 240 echo chambers.

The Edit Room, stores a ½-inch Studer A-80 RC stereo 2-track machine, and a bank of six Sony 700ES cassette decks. A Magnatech film dubber, 3M ¼-inch ¼-track machine, and a 3M ½-inch mono machine are also situated in The Edit Room.

TRANSITIONS FROM OLD TO NEW

Robbins spoke of the transition period and its lack of effect on business as usual. "There was an opportunity for the 'old' Sigma Sound to book a session with Joe Cocker. We decided that if we could make the transition smooth, we could best serve the client. So we took provisions to do just that in terms of staffing, maintenance, and general administration. The end result was that the session started about a week before we took over, and ended about two weeks after we took over. The producer, Charlie Midnight, was aware that there were changes being made, but we tried to keep that as transparent as possible, and they had a very good session."

Robbins, whose background spans the producing and directing realm, views the acquisition of Sigma as being a "tremendous commitment for us and myself personally. I think it's a wonderful, wonderful challenge." In looking toward the future, Robbins maintains an optimistic view as to the direction Sigma is headed in.

"Our goal, obviously, is to make Sigma as successful as possible. We want a healthy market share, so we are taking steps, in terms of the technology and staffing of the facility, to insure that occurs. We're coming into an excellent facility, and we want to continue to make it grow." His philosophy solidifies his view: "You take a good thing and try to make it better. You give the best service you can to a client, try to provide an atmosphere that's best for them to work in, and provide a situation for them whereby they will want to return."

EQUIPMENT LIST

STUDIO A

Monitors:

Fostex 3-Way
Little Reds in Studio

Amps:

Crown PSA-2
Hafler P 225
Red Series Time/Sync Crossover
Marantz

Console:

SSL 6000E 40-input
SSL E-Series Computer w/ Total Recall

Audio Machines:

Studer A800 MR III 24-track
Ampex ATR-100 ½-inch 2-Track
Ampex ATR-100 ½-inch 4-Track
Ampex ATR-100 ¼-inch 2-Track
Nakamichi MR-1 Cassette
Technics SL-1500 Turntable
Reverbs:
EMT 140 Mono Plate
EMT 140 Stereo Plate
EMT 240 Stereo Plate
Yamaha SPX-90
Roland SRV-2000
Delays:
2 Lexicon PCM 41
2 Lexicon PCM 42
Lexicon Prime Time 93 D
Compressors/Limiters/Levelers:
Drawmer 1960
Harmonizer:
Eventide H 910
EQs:
API 553 (8)
2 Pultec EQP-1A3
dbx De-esser 902
Noise Reduction:
Dolby M 24H
Synchronizer:
3 Lynx
Instruments:
Yamaha 5-piece Drum Set
Steinway Grand Piano

STUDIO B

Console:
Neve VR 60-input with Flying Fader automation
Monitors:
Big Red Monitors w/604E2s and UREI Horns
Little Reds in Studio
Amplifiers:
Crown PSA-2
Crown DC 300A
Red Series Time/Sync Crossover
Noise Reduction:
Dolby M24H
Dolby 2 Track Record and Playback Capabilities
Synchronizers:
2 Lynx Time Line
Video:
JVC ¾-inch deck
Barco Monitor (Large)
Reverbs:
EMT 140 Mono
EMT 140 Stereo
EMT 240 Stereo
SPX-90
Delays:
2 Lexicon PCM 41
2 Lexicon PCM 42
EQs:
API (4)
550A (4)
553
2 Pultec EQP-IS
Misc. Aux. Gear:
dbx 902 De-esser
Drawmer 1960 Compressor/Limiter
Eventide H910 Harmonizer
Teletronix LA-2 Comp/Lim
Audio Machines:
Studer A 800 MKIII 24-Track with Remote
Studer A 820 ½-inch 2-track
Studer A 820 ¼-inch 2-track
Nakamichi MR-1 Cassette Deck
Ampex ATR-100 ½-inch 4-track
Instruments:
Yamaha 5-piece Drum Set
Yamaha Grand Piano
Extra Audio Gear:
Technics SL-1500 Turntable
Pioneer PD91 Compact Disc Player

STUDIO C

Console:
SSL 6000E G-Series Computer (E-Series Capability)
56 Inputs (including 8 Stereo Modules)
Monitors:
Big Red Speaker Cabinets w/604 EZ and UREI Horns:
Little Red in Studio
Red Series Time/Sync Crossover
Amps:
Crown D150 A
Crown PSA-2
Hafler-P225
Marantz-250M
Audio Machines:
Studer A800 III 24-Track
Studer A820 ½-inch 2-Track
Ampex ATR-100 ¼-inch 2-Track
Tascam 122 MK II Cassette
Technics SL-1500 Turntable
Reverbs:
EMT 140 Mono Plates w/Remote
EMT 140 Stereo Plates w/Remote
Lexicon 200
Yamaha SPX-90
EQs:
2 Tube Tech
4 API 550A
2 dbx De-essers 902s
Compressors/Limiters/Levelers:
dbx-165
Teletronix LA-2A
Noise Reductions:
Dolby 361s
Dolby M 16Hs
Delays/Harmonizers:
Eventide H910
Lexicon Delta-J
2 Lexicon PCM 41
2 Lexicon PCM 42
Synchronizers:
3 Timeline Lynx Timecode Modules
T.V. Monitor:
Barco Large 21 inch (approx.)

FRANK SERAFINE

Creative Use of the Macintosh for Audio Post Production

For the past decade, many "scoring & post" facilities have been synchronizing multi-track tape recorders with videotape using SMPTE time-code interlock. This type of electronic editing was used on a film as early as Walt Disney's *Tron*. Although this was an improvement from moviola-type systems, some of the simplest tasks performed on 35 mm magnetic tape are simply not possible using multi-track tape, for example, easy slipping of individual tracks. In addition, synchronizer lock-up times can be very time consuming. New advances in hardware, combined with off-the-shelf computers are creating a new kind of tapeless studio. A tapeless studio combines the speed of multi-track with the flexibility of mag and first generation digital sound quality. As we look forward into the 1990s, the tapeless studio is becoming a moderate-cost reality.

Since the Macintosh computer was introduced four years ago, it has evolved into a powerful system and is now considered the *de-facto* standard of the entertainment industry. A major factor in this popularity of the Macintosh is its ease of use. Most programs can be run by non-technical personnel with little or no training. In addition to the short learning curve, there is a wide selection of software available. Finally, its portability allows the user to work at home if so desired.

MIDI INTERFACE

The MIDI interface is the connection from the computer to the outside world. MIDI is a serial interface standard, which is the link between a computer and digital samplers, synthesizers, and signal processors.

Opcode Systems studio & box acts as a SMPTE to MIDI time code (MTC) converter in addition to being a MIDI interface. SMPTE can be longitudinal (LTC) and recorded directly onto the analog track of a tape or embedded into the vertical interval (VITC). VITC time code is displayed more accurately at slow speeds and is useful for logging and editing sound effects with video tapes.

Some software packages capture time-code numbers on-the-fly, which eliminates most typing and allows more time for creative applications.

Serafine Inc recently composed music for the 1990 Chrysler commercials. To synchronize the music we used Opcode Systems new Vision sequencer package with their time-code interface.

MIDI control has allowed digital samplers, such as the Emulator III, to be an effective tool for sound design and music production. Digital sampling devices store sound by digitizing and storing stereo sounds from a line source or microphone.

The "sounds" storage medium has just recently become available on the market, for example: the Emulator/Syquist 45 megabyte removable storage system. For fast recall of very large music and effects libraries, Optical Media offers a 550, and the Sony Pinnacle REO 650 megabyte read/write erasable removable-cartridge-drive device. The ability to perform-to-picture is an advantage to sound designers because of the expression and control over the sound. Samplers differ from hard disk-based recorders in that they are of the variable-rate system. This means that they are good for sounds which require rapid playback and transposition of pitch.

Figure 1. Serafine's arrangement of Debussy's Clair de Lune is completely orchestrated in this Vision Graphics Editing Window. Eighteen individual synthesizers and samplers are controlled.

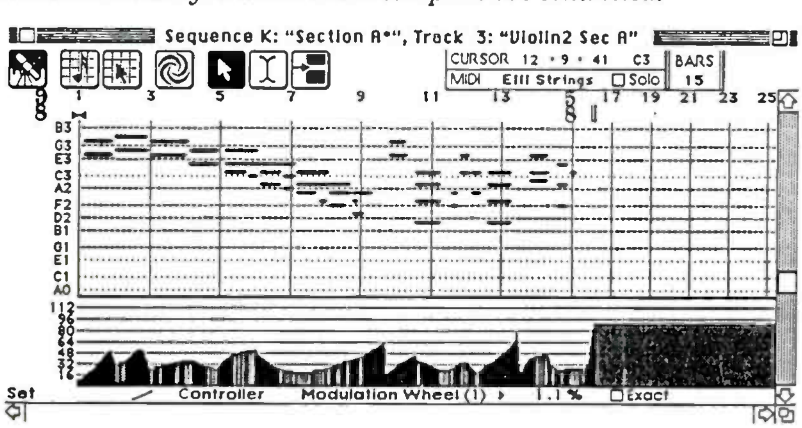
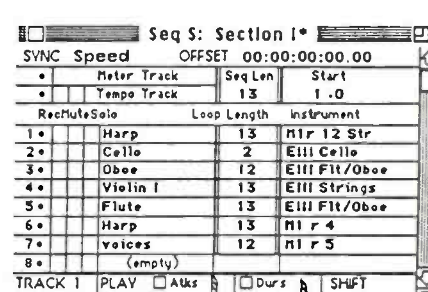


Figure 2. The Vision Sequence Window. Again, Clair de Lune and showing the arrangments of instruments, channels, tracks; SMPTE and MIDI information are handled.



Another advantage of the Macintosh-based studio is that there is no clear cut division between recording, mixing, and editing—the user can be mixing and editing at the same time using MacMix software from the Dyaxis IMS.

This is a departure from film-based methods. Furthermore, the Macintosh studio has a flexible, open-ended approach, meaning it works with different types of hardware which use the Macintosh as a "front end." Using software such as "FAME" from Fostex, a user can automate punch-ins and punch-outs on a conventional multi-track recorder connected to a Fostex synchronizer.

HARD DISK

A hard-disk recorder writes a stream of digital bits onto a spinning disk drive. Because it is random access in nature, it has none of the long shuttle and locking times of conventional tape recorders. With Q-Sheet A/V from Digidesign, a digital audio processor, and a hard disk such as Dyaxis, the user can rapidly assemble music and sound effects, slip tracks, and even edit dialogue. In a recent Anderson Consulting spot "CEO/Times Square" for Y&R (New York), we designed the soundscape for Times Square on New Year's Eve in the year 2000. The main challenge of the spot was the creation of a huge crowd which counts down the descent of the ball. A group of about fifty people were recorded using a portable digital recorder, and assembled and mixed digitally using the Dyaxis hard-disk system. Hard-disk recorders allow multi-track recording, editing, mixing, and mastering, all within the digital domain for maximum fidelity.

SOFTWARE

Alchemy 2.0 from Blank Software is a visual sample editor for the Macintosh. A sound can be analyzed for harmonic content and the computer displays frequency response, phase and level over many bands, which can then be adjusted and re-synthesized to reflect a new spectral content. Sample editors also perform non-real time processing such as time compression, automatic tuning, digital equalization, and more. Furthermore, actual re-drawing of a waveform by hand is possible in high-resolution display mode.

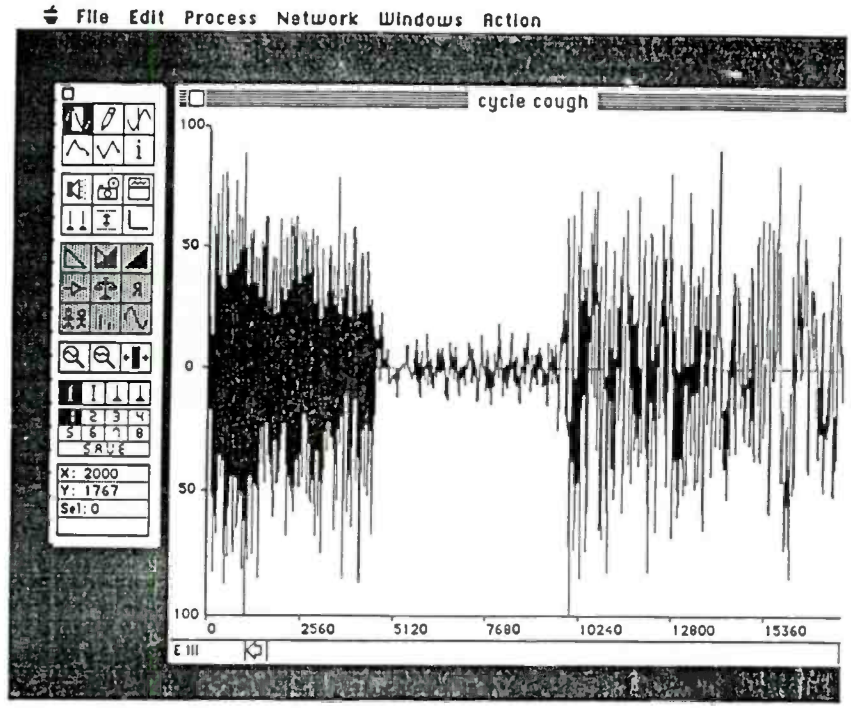
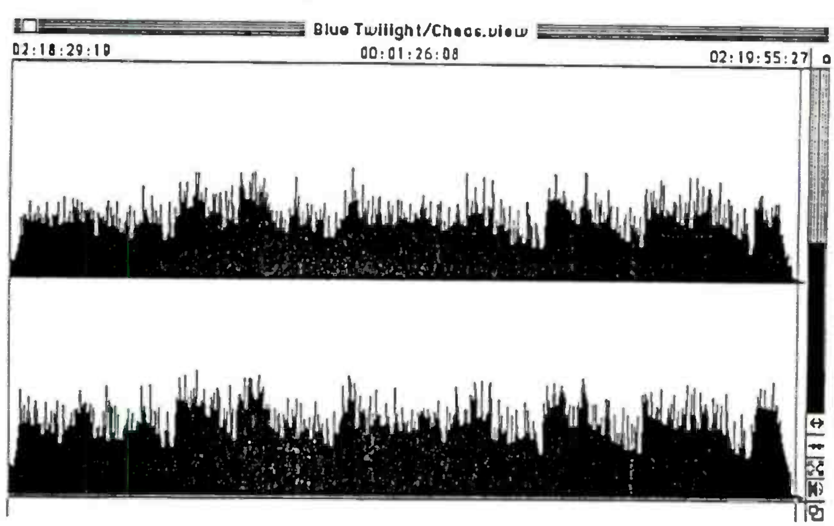


Figure 3. The Alchemy Sound Sample Window displays a sound camera that graphically shows waveform and amplitude against time.

Figure 4. With Finale Music Notation Software, music can be printed, alterred, and re-arranged—all on screen and through sequencer MIDI files

Figure 5. The Dyaxis Sound File Window. Dyaxis can store 1-½ hours of digital stereo information—controlled through time code and internally looped and otherwise manipulated.



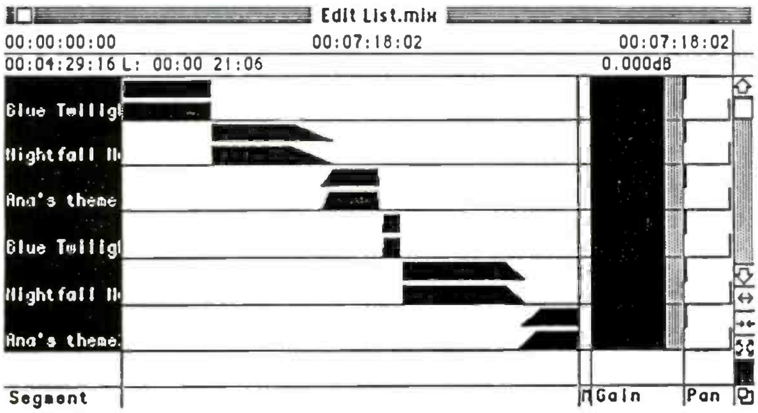


Figure 6. Dyaxis Edit Window where music can be edited, mixed, and even crossfaded.

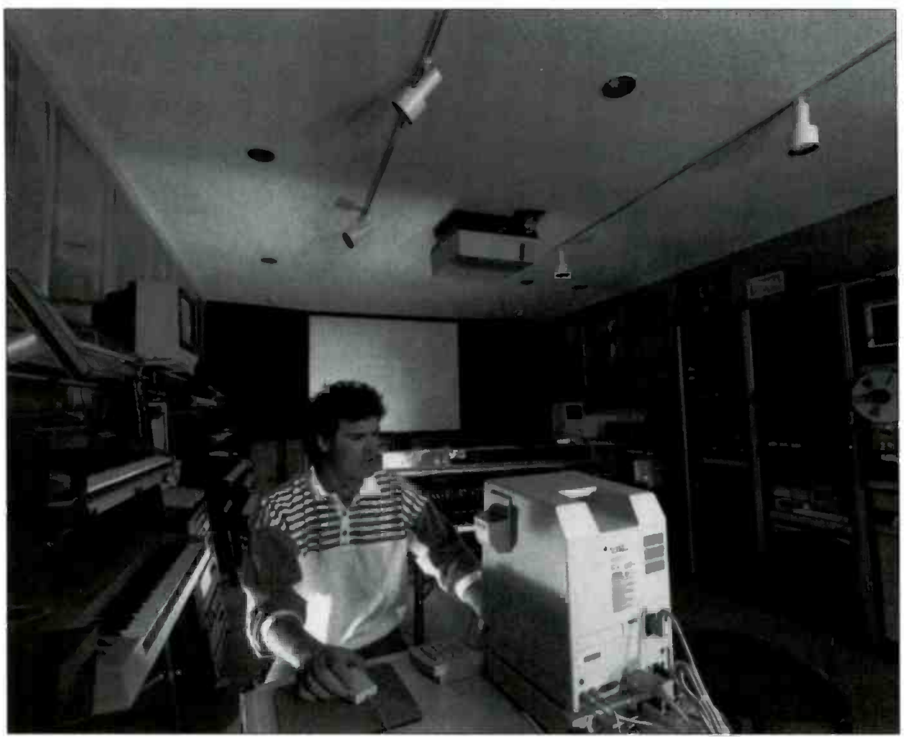


Figure 7. Frank Serafine at the Macintosh in his studio.

Q-sheet A/V is a SMPTE-locked MIDI automation program for the Macintosh. In addition to event triggering, it controls external VCA based fader automation packages and signal processors. Time-code locations can be captured into the program in real-time and trimmed or triggered directly from an existing CMX-style EDL. Q-sheet A/V was used to edit all of the background and special sound effects for *Life on the Edge*, a soon-to-be-released feature about a society designed by plumbers.

CONSOLE AUTOMATION

MIDI is also used for console automation with a Macintosh software package. Total recall of signal processors during the mixdown process enables extremely precise control of effects and equalization with memory of all settings. This automation technique was used to precisely manicure the robot ADR tracks on *Short Circuit II* (Tri-Star Pictures), which was recently nominated for best ADR editing by the Motion Picture Golden Reel Awards.

The Kurzweil Master MIDI controller acts as an intelligent interface between computers, keyboards, and software. Hundreds of intricate programs can be set up to operate, function and control the entire MIDI studio. Soft-keys can be a multitude of different functions; for example, remote control of an external MIDI controlled reverb device such as the Lexicon PCM-70. MIDI control of signal processors during the mixdown process enables extremely precise control of effects and reverb with recall of all settings. For example, with the use of a MIDI controlled reverb, room sizes would change to correspond with scene changes.

One final important feature of a computer-based studio is that it allows multiple users to share files over a local area network (LAN). Multiple workstations are made possible using local talk hardware which is built into every Macintosh. Remote work stations can be connected using normal phone lines and a low-cost modular adapter.

With a central hard disk and a file server, sound editors can literally work out of a small room sharing a common sound library, with little more than a Mac and a video monitor. Also, sound edits can be performed on the Macintosh and auditioned using its internal speaker.

SUMMARY

The Macintosh-based studio has many advantages over traditional systems. Unlike tape machines, there is no alignment and little maintenance. Unlike analog systems, there is no added hiss and no generation loss. Since the computer remembers all settings, there is maximum flexibility to make changes easily.

The Macintosh-based studio reads SMPTE time code and locks to picture. Ideally, every device in the studio is interconnected and talks together—exchanging SMPTE, MIDI, VITC, SCSI, and computer network data. Old skills are not abandoned, but applied using new hardware and technology. Since the Macintosh interfaces with older technologies, such as tape machines, a facility is not forced to convert to everything at once, but can mix and match to use the best of both worlds. db

ARLON OBER

The Changing Face of Film Scoring Using the "New Technology"

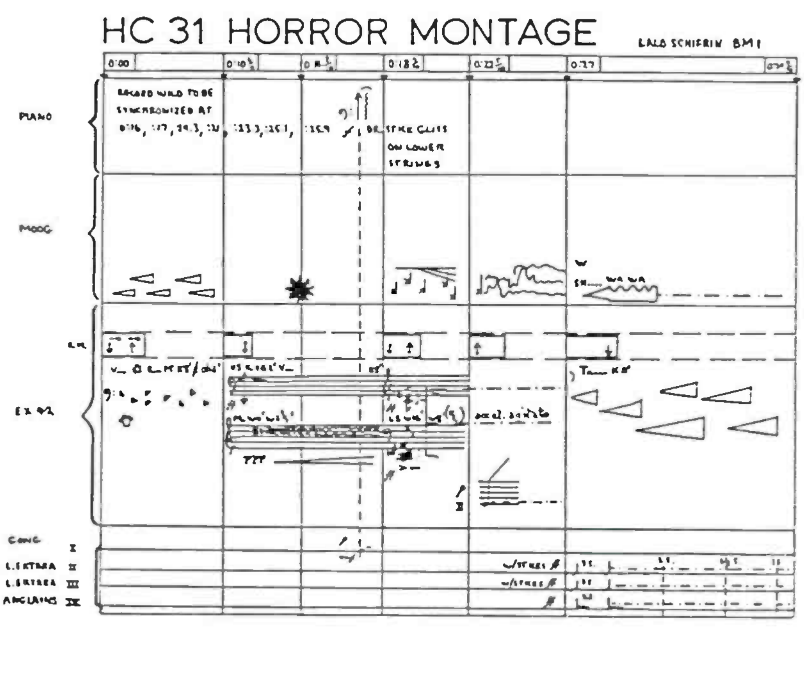
The author begins with a short history of film scoring

When I was asked to write this article on contemporary film scoring techniques, I began thinking of poor old Rip Van Schwartz who used to sit at the Great Organ in Radio City Music Hall and re-create renditions of the classics and invent on-the-spot moods for the great silent films. It seems he fell asleep one day at the organ and woke up just the other day in Los Angeles at a MIDI-equipped synthesizer studio and immediately went into shock. The truth is that Rip could have fallen asleep in 1978 and still gone into shock. The state of scoring music to picture has changed so much over the years—dramatically so in the past decade alone—that many composers brought up learning just harmony, counterpoint, and orchestration techniques are being left in the dust as the new technology begins potty-training.

In spite of any romantic notions we may have about the beauty and drama of music, it has almost always been a functional art: to accompany Greek dramas, to ease the dissemination of Church liturgy, to impress and soothe the royal European courts, to dance to...and, ultimately, as entertainment in its own right. In the noisy little theaters that sprung up at the beginning of the century to show 2-reelers, music served other—if somewhat less lofty—functions: hide the projection noise and keep the audience from fidgeting. Film producers felt the audience would watch the picture if they were *hyped* by emotional music.

The first music played for pictures was often just gleaned from the romantic *classics* of the preceding 75 years. Such books as *Motion Picture Moods* by Erno Rapee were available from G. Schirmer and other reputable publishers containing the works of Grieg, Mendelssohn, Beethoven, etc. The pianist would choose material by categories, such as *Orgies*, *Oriental*, *Parties*, or *Passion* and "wing-it" as the picture flew by. By the early 1920s, an occasional score would be created by "serious" composers such as Milhaud, Honegger, and Shostakovitch for a major motion picture. Although, in some big city theaters these scores were often performed by full orchestras, elsewhere a pianist or organist like our friend Rip would perform it—often having never seen the film.

Figure 1. A page of score from "The Hellstrom Chronicle," music by Lalo Schifrin.



With the advent of "talkies," the techniques changed. Music was timed to the scene and recorded to the projected image with a live orchestra onto monaural optical film. Over the years, techniques such as the click-track and picture streamers were invented to allow great precision in hitting cue points in the film. By the 1940s, such composers as Carl Stalling could hit the chomp of Bugs Bunny's carrot with a nasty chord if he so desired. Serious film composers such as Alfred Newman, Franz Waxman, Max Steiner, Bernard Hermann, and Miklos Rozsa—to name but a few—were creating and producing dramatic original compositions for the cinema with 100-piece orchestras in massive studios. All were recorded to picture using a hefty staff of union musicians, recordists, and technicians. Hefty, too, was the cost. But the cinema had be-

"PARATROOPER" Music cues for Picture REEL 6

6M1 "KIM FINDS VAN" SYMPTE Start: 6:00:08:03
Scene: Music in on cut, Kim stalking area with camera
Music ends on fade out prior to 6:00:48:11, cut to Rayker cleaning knife.

6M2 "KIM GETS THE FINGER" SYMPTE Start: 6:02:28:00
Scene: Music in as Kim begins to realize she's holding something weird in her pocket.
**Note: align cue so that all cabin cutaways & running scenes coincide with music changes.
Music ends on Kim hiding, fade out French horn on cut back to cabin. (6:04:10:17)

6M3 "SLOW LEARNER" SYMPTE Start: 6:04:29:00
Scene: Music in on cut to Major after Rossi's knife gesture - low strings.
Music ends/segues at 6:05:27:15, soldiers on path.

6M3A "ON THE ROCKS" SYMPTE Start: 6:05:27:15
Scene: Music in on cut soldiers on path (with drums & brass).
Music ends at 6:07:48:15 as Hara rifle fires-FX in clear.

6M3B "IT'S A GAS" SYMPTE Start: 6:07:51:15
Scene: Music in scene as Rossi at van reacts to explosion.
**Note: align cue so that at 5 seconds into cue Horns & Low Strings hit cut to Bates.
Music ends on hold & fade out as soldiers arrive-Cooper kneeling by Rossi (6:08:56:15).

6M4 "KINGDOM COME" SYMPTE Start: 6:09:24:15
Scene: Music in on cut to smoking grenade launcher on ground.
Music ends by 6:09:46:12 as reel ends.

CHH 51.40 Background, Instrumental ✓ 6:14.4
CHH 40.32 Background, Instrumental #6
CHH 1:42.63 Background, Instrumental
CHH 53.60 Background, Instrumental
CHH 2:21.00 Background, Instrumental
CHH 1:05.00 Background, Instrumental
CHH 21.83 Background, Instrumental / 7:09.46
CHH 2:37.17 Background, Instrumental #7
CHH 2:15.04 Background, Instrumental
CHH 1:05.60 Background, Instrumental
CHH 56.16 Background, Instrumental 6:52.77
CHH 1:17.40 Background, Instrumental #8
CHH 53.32 Background, Instrumental
CHH 1:42.63 Background, Instrumental
CHH 1:20.40 Background, Instrumental
CHH 1:02.24 Background, Instrumental

VILLAGE FACEOFF(7x5)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	2:37.00 Background, Instrumental	/ 9:07.04
BULL IN THE VILLAGE(8x1)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	1:10.32 Background, Instrumental	#8X
THE BIG CHAT(8x2)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	2:13.40 Background, Instrumental	
HANDRUN(8x3)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	3:15.92 Background, Instrumental	
CAVALRY ARRIVES(8x4)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	2:16.00 Background, Instrumental	
DOG TAGS(8x5)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	11.32 Background, Instrumental	/ 9:14.96
SAYING GOODBYE(9x1)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	1:45.04 Background, Instrumental	#9
REALIZATIONS(9x2)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	41.00 Background, Instrumental	
DEATH BELL(9x3)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	2:31.04 Background, Instrumental	
TROPHY(9x4)	ARLON OBER	100	ALTAVIVA-LOGO	CHH	1:40.84 Background, Instrumental	/ 6:31.60

TOTAL MUSIC 1:18:12.37

Source Q's: 1:07.44
1:31.04
6:09.25
1:29.48

Total 4:17.61
1:18:13.72

PARATROOPER/ Page 2

Figure 2. A page of the master cue list from "Private War" (a.ka. "Paratrooper").

come the great American escape, and the modest price of an admission ticket guaranteed two hours away from the daily grind. A lucrative new industry flourished.

A NEW MEDIA ARRIVES

When TV popped its little head in the door in the late 40s, music was done live from the broadcasting station, much the same way radio had been done up to then. Networks often carried a house orchestra of 50 to 60 players. Short play-ons and play-offs were performed for each show—live. It became a selling point to hear *The Jack Benny Show* accompanied by the *NBC Orchestra*. In the late 50s and early 60s, filmed episodic TV grew up along with longer forms of filmed entertainment, and all of the cinematic techniques of movie scoring were incorporated, though the orchestras were scaled down to fit the status of TV at the time. Music was written, copied and played by live humans and recorded to projected image. But a new horror was waiting just around the corner—The New Technology.

THE NEW TECHNOLOGY

It began rather innocuously in the 1950s with the electric guitar, bass and piano and by the early 1970s had grown into big, complex gadgets: multi-track recording, 2-inch video tape that had to be edited with a razor blade, the Moog and Emu synthesizers that looked like spaghetti explosions hanging from an Edison nightmare, the Buchla synthesizer that looked like an electronic terror in a suitcase. Very scary! There were new terms for the composer to learn: analog, filters, VCAs, VCOs, wave forms, pink and white noise. Creating sounds might take hours of experimentation. Notation for these sounds was anybody's guess. (See Lalo Schifrin's score to the *Hellstrom Chronicles*, *Figure 1*.) To compose a film score using all synthesizers was tedious and immensely

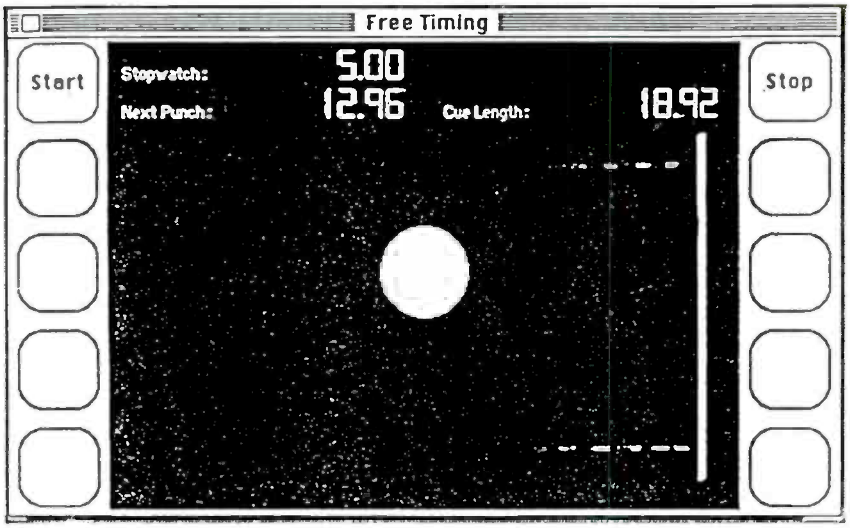


Figure 3. Above, Free Timing page on the cue program. Figure 4. Below, Music notes and graph from "Private War" (a.k.a. "Paratrooper").

09:04:48:08	1:11.00	CUT	MS, R falls back	40-	3.50
09:04:49:08	1:12.00	CUT	CU, C reaches down to punji stick...	41-	1.73
09:04:51:00	1:13.68		ss, pulls stick out here	42-	1.48
09:04:51:10	1:14.08	CUT	MCU, C turns and swings	42-	2.38
09:04:52:17	1:15.36	CUT	CU, R gets it in gutlll, Yells, falls back	43-	1.23
09:04:53:12	1:16.16	CUT	MS, R staggers back to CAM	43-	3.03
09:04:55:00	1:17.68	CUT	RUV MS, R staggers against wall...	44-	2.42
09:04:57:00	1:19.68	CUT	MS C jumps up... grabs rafters, ...+	45-	2.89
09:04:58:04	1:20.84	CUT	MS,C kicks R in face	46-	1.47
09:04:58:16	1:21.32	CUT	CU, R falls against wall...C runs in...	46-	2.55
09:05:00:16	1:23.32		ss, pins R against wall, HOLD MX HERE,	47-	3.02
09:05:03:08	1:26.00		ss, R whispers; "THIS IS WHAT IT'S ABOUT...DEATH IS YOUR ALLY...LOOK INTO MY EYES KID, WHAT DO YOU SEE?"	49-	1.00
09:05:13:00	1:35.68		EOL, C looks back and says	54-	2.61
09:05:13:16	1:36.32		"MY ALLY SARG"	54-	4.03
09:05:15:16	1:38.32		EOL, C reaches up...	55-	4.50
09:05:18:12	1:41.16✓	CUT	CU, C grabs R's left hand, as he pulls knife out from wall	57-	2.84
09:05:19:12	1:42.16	CUT	CU, C pulls knife down...reaches up	58-	1.08
09:05:21:00	1:43.68✓	CUT	ECU C slices R's fingers offlllllllllllll	58-	4.47
09:05:22:12	1:45.16	CUT	CU, R screams and sinks to floor...	59-	3.77
09:05:29:12	1:52.16	CUT	ECU C lauds over R..HOLD CHORD HERE	63-	3.39
09:05:32:08	1:55.00	CUT	CU, Pics of R and C's dad in NAM	65-	1.73
09:05:34:00	1:56.68		ss, R VO; "GO ON COOP,"	66-	1.48
09:05:36:00	1:58.68	CUT	EOL, CU R; "YOUR MOVE COOP"	67-	1.95
09:05:37:21	2:00.52	CUT	EOL, ECU C looks...	68-	2.00
09:05:39:14	2:02.24	CUT	CU, R says: "YOU'RE WORTHY NOW"	69-	1.90
09:05:41:16	2:04.32		EOL, R looks up ...Looks down	70-	2.55
09:05:44:16	2:07.32	CUT	CU C Looks at wall	72-	1.24
09:05:47:00	2:09.68	CUT	CU Photo...	73-	2.51
09:05:49:08	2:12.00	CUT	CU R sobs "Come back Coop,"...	74-	3.69
09:05:51:08	2:14.00		EOL, ...R bewildered...adds	75-	4.16

PARATROOPER 9x3 Page 3 8/29/88 2:57 AM

time-consuming, and synthesizers often merely became an additional color to the conventional orchestra. No two machines looked or functioned quite the same way, and each had its own unique characteristics. To create film music on these analog dinosaurs, you needed at least three things: a lot of time, a lot of money, and oodles of imagination. Then, at the beginning of the 1980s, a whole new set of breakthroughs occurred.

When chips replaced circuit boards, which had replaced tubes and wires, a series of products appeared which have grossly altered the music scoring scene in this decade alone. Among these: FM synthesis-machines such as the DX-7 series, the *inexpensive* personal computer, the SMPTE standard code, miniature multi-track recorders, digital samplers and processors, rack-mounted sound modules, digital sequencers, and, most of all, MIDI. Film composing had reached a fork in the road and like in the Frost poem, a major decision had to be made: which way to go?

A CHANGING APPROACH TO FILM MUSIC: MARKETING VS. FUNCTION

Technology was not the only thing changing. While TV was growing up and cutting into the immense power of the Hollywood studios, the politics and the *business* of show business was changing. As movie audiences stayed at home in increasing numbers to watch their favorite network shows and union wages rose exponentially, Hollywood executives felt the crunch of increasing costs and diminishing returns. New avenues were being explored to re-attract their audiences and increase their revenues. Songs in films, which had always played a major role in reinforcing a film's image as well as gathering additional income for the Hollywood studios (who had long since entered the publishing business) were taking on a new impact. Films became sales grounds for launching new songs, records, and even books. Cross-marketing became a new weapon to fight the new scourge of network and cable TV. As the squeeze for the viewer bucks increased, all film and TV media began to take new approaches to the marketing strategy: pinpoint the buyer and gear the product to what he or she wants. Maximize the profit

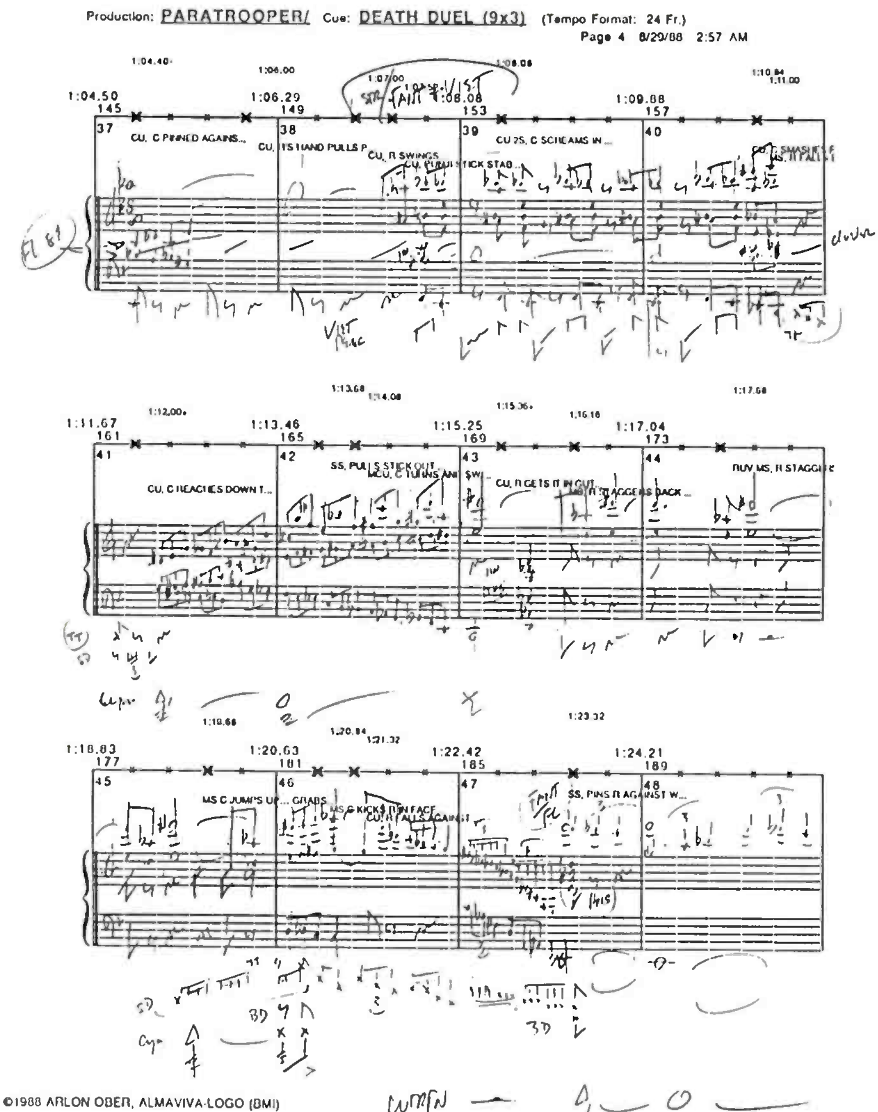


Figure 4(A). A sketch page from "Private War." (a.k.a "Paratrooper").

through saturation sales of all possible related products. What? Okay, in plain language: the movie industry will sell anything for a buck.

Today, not only will a film studio sell all film-related ancillary products (records, cassettes, CDs, videotapes, books, paperbacks, comics, T-shirts, and lunch pails), but they'll even make a film based on a hit tune (*Tallahatchee Bridge*, *Take This Job and Shove It*). Luring a teen audience via rock-laden soundtracks—with all its potential sales points and reinforcement through free MTV renditions—has forced mood music to take an ever-diminishing secondary role in pictures. Even dialogue scenes, characteristically played with underscore, may now be covered by loud vocals and inappropriate lyrics if test marketing warrants success. The film *Coming Home* was the first to insistently cover dialogue with music, but the technique has come into increasing usage, as exemplified by such shows as the now defunct *Miami Vice*. Blatantly emotionless (though successful) techno-pop as in Stewart Copeland's scores to the *Equalizer* TV series has also made a considerable sociological statement about the emotional indifference of today's urban society.

A CHANGING SCENE

The increasing emphasis on incorporating commercial rock and other forms of popular music into scores has given rise to two other contemporary entities gleaned from the record industry: the *new breed* of Music Supervisors, and the "new breed" film composer such as the incredible Danny Elfman of Oingo Boingo, and Mark Knopfler. In past decades, *Music Supervisors* functioned as composer-studio liaisons and primarily dealt with the choice of composer, the spotting sessions which decided music placement, the contracting of musicians, and supervision of the recording along with clerical functions of contracts and clearance forms. Also, they were often expected to locate material or songwriters to "bookend" the film with a relevant "pop" song or songs of the period. The Music Supervisors often function exclusively as negotiators in choosing and clearing existing record cuts.

The new breed film composers are often chosen because of their draw or appeal to a young audience, yet are most often asked to create a conventional—albeit "hip"—musical score. Traditional film composers are now expected to do both conventional and new forms of scoring including synthesized scores and rock. Maurice Jarre, who composed the orchestral scores to *Laurence of Arabia*, and *Dr. Zhivago* has recently turned out a number of totally synthesized scores for such pictures as *Witness* and *No Way Out*. The synthesizer-orchestral score has also found a niche. Arthur B. Rubenstein's score to *Blue Thunder* (all synth with live brass), and Joe Renzetti's score to MGM/UA's *Child's Play* (all Synclavier with live orchestral overdubs)—which incidentally, I orchestrated and conducted—are prime examples of the latter. Increasingly, though, the all-synthesized score has crept into the limelight.

Partially due to low budget, or the increasing costs of live scores, or the attempt to utilize the newest sounds, the all-synthesized score has become a valid entity and is not going to go away regardless of the popularity of pop music. The ability of samplers and analog-digital synthesizers to reproduce conventional acoustic instruments as well as new, innovative electronic and unheard of acoustic-type sounds has established the validity of this new medium. The advent of MIDI, the inexpensive personal computer, the evolvement of the drum machine, the vocal synthesizer and miniaturization of multi-

Figure 5. The synthesizer room at the Lambert Studio.

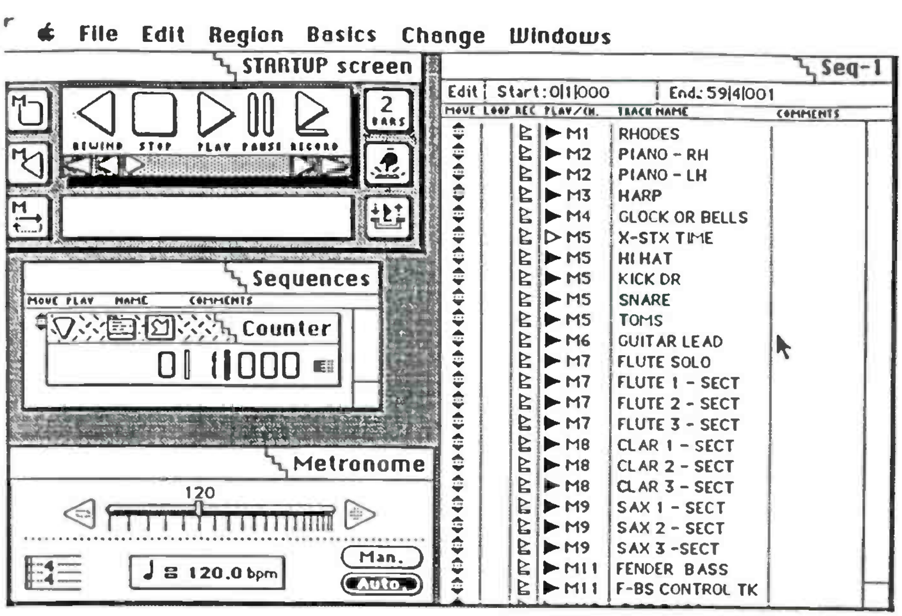
track machines in a variety of new formats has created a new type of recording procedure based in the "home studio." Now, even demos can be finished products, complete with vocals, brass, and strings. Many producers look upon this new wave of technology as well as the all-song soundtrack as panaceas for their musical needs. This is not necessarily the case.

THE TIME FACTOR

One major fallacy in requesting a synthesizer score is the public's lack of awareness as to the length of time it takes to create and produce music this way. There are basically two ways to create synthesized scores. The first is the "winging-it" technique of syncing up a multi-track tape recorder, a computer-controlled synthesizer and a videotape of the film by way of SMPTE code through a sync-box like the Roland SBX-80. The composer watches the picture several times and plays along with it—actually a lot like our friend Rip Van Schwartz. In additional passes, he may add numerous other parts, building up the texture as he goes. "Dead hits" can be adjusted later on the computer and brought into exact sync via SMPTE time code. For fairly abstract or sequenced material this technique is practical, but for conventional scoring using melodies, counterpoint, and harmonies, it can become chaotic unless one has a computer-like memory and, preferably, perfect pitch. Using this technique, a composer can produce a 2-minute 30-second cue in anywhere from 3 to 8 hours depending on the speed and complexity of the piece and the facility of the composer. He has avoided all the steps of scene breakdown notes, paper notation of the music, orchestration, and recording procedures. Not bad, eh?

The second technique is the use of synthesizers to *realize* or perform material that otherwise would be recorded by a live orchestra or band. This is certainly a slower procedure than winging-it, but the results are far more controllable and orchestral writers will perhaps relate to it more. It usually heaps many additional chores onto the composer, and the results are often lacking in the subtle sonorities and rich textures of a live performance. Let's go through a step-by-step run-through of composing a cue with the state-of-the-art technology.

Figure 6. The "startup" template on the Performer program.



GETTING STARTED

Today's composer starts his work with several meetings with the producer, the director, and the music supervisor. After an initial meeting to discuss style and budget, a music editor is often present. The film is reviewed thoroughly and placement, highlights, moods, and hit points are discussed. The most important thing in this meeting is the start and stop points of each cue and the creation of a master list of cues. The master list of cues should always include the starting and ending points in either *feet* and *frames*, or in SMPTE code numbers where "00" or "00:00" is *always* the start of each reel from the *Academy Leader* (8 seconds from the first frame of picture) (*Figure 2*). Each cue should indicate at the very

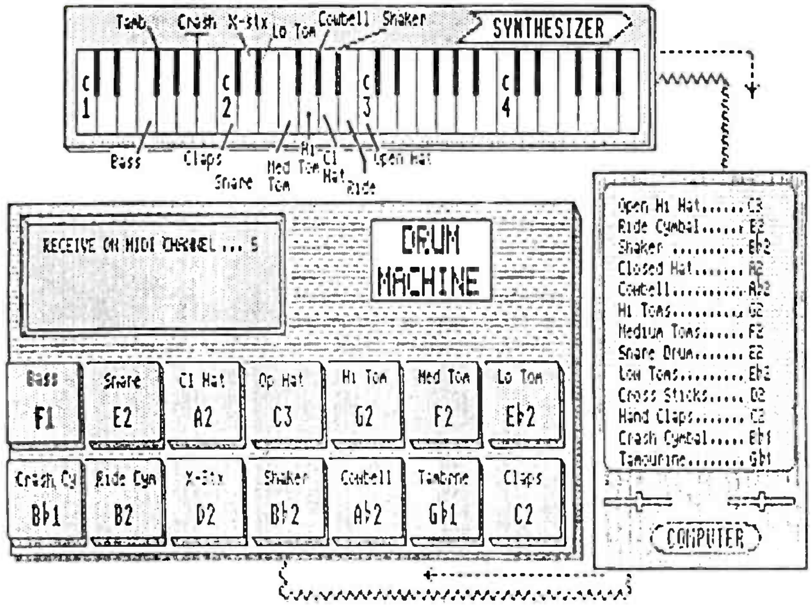
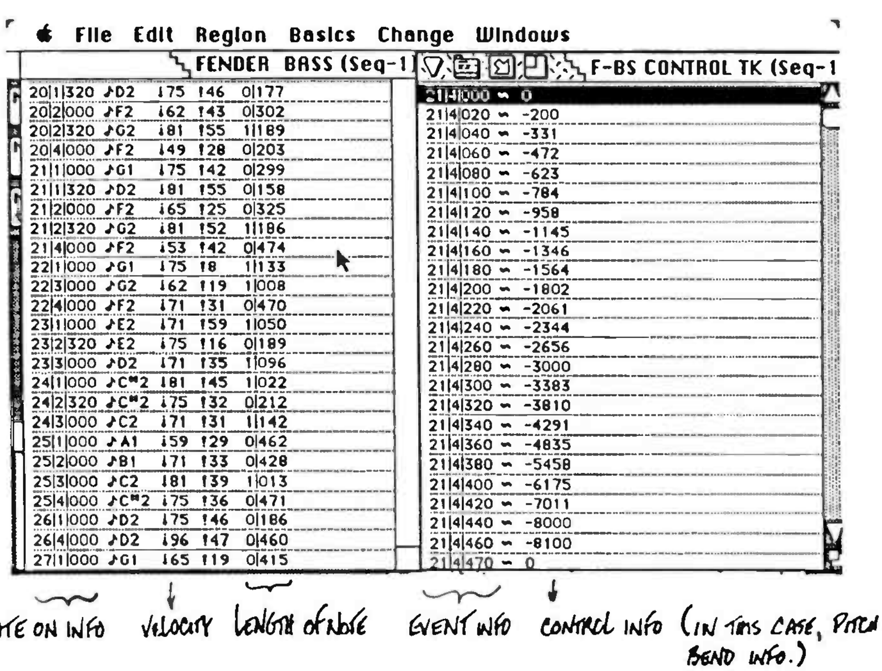


Figure 7. Triggering a drum machine from the synthesizer.

top of the notes where in the reel it starts (either footage or SMPTE #) and that number usually becomes the "00" or "00:00" of the cue. All points inside the cue must be referenced to that number. By keeping the integrity of the cue intact, no changes will be needed in the event of film changes elsewhere. Likewise, if timings *inside* one cue are changed, the integrity of the rest of the cues will remain intact.

I personally like to have music notes entered into a computer program such as "Cue" from OpCode, available for the Mac. All timings may be dumped directly into the program from the SMPTE track on the video, which saves a great deal of time typing 4 or 5 digit numbers and colons. Descriptions still must be inserted, but appropriate abbreviations can help out. Boxes for "Cut" and one for "Key Hit" (important events) are present besides each number and may be triggered from mouse or keyboard. On the printed cue sheet, the word "Cut" will be inserted in front of scene changes and a *checkmark* will be inserted for any "Key Hit." Once all information is inserted, the program will search out tempos for choices of click or metronome number, and show a listing of all hit points in any given tempo. It will also give you the options of retarding or accelerating a section or sections to hit at any given point. Very slick!

Figure 8. At left, fender—bass track. At right—bass control track.



A wondrous aspect of this program is the "Free Timing" mode which shows you a black screen and sets off streamers and punches to coincide with the opening of the cue, each "Key Hit" point and the ending (*Figure 4*). When the cue is adjusted to your satisfaction, a print option allows you to screen or print the actual cue information inserted over blank music staves. Scoring paper may be created of up to 12-4 bar staves per page. This process may save about 30 minutes work alone on each cue. On a feature score with 40 cues, this can amount to saving a valuable 20 hours of time. Changes or revisions are no longer a nightmare. Even if you decide to wing the cue, this program is most helpful. There are now services that will do all the note-taking and typing, and simply telephone you the information for your "Cue" program via modem.

Once the music notes are completed, an additional review of each cue is most necessary to flesh out the style, mood and flow. I have developed a technique of *graphic* visual cues which I like to notate beside the written information. These cues are really helpful in writing while away from the picture, and may include tension levels, pyramids, dynamics, wires, bass lines, themes, pauses, and holds (*Figure 3*). Composing the music and the creation of synchronous mood sections is something I am not going to discuss. Every composer has his own techniques, instincts, and approach, and many different results may still be appropriate. What I would suggest, however, is that each composer consider the music-to-be as "The Effect of?????" rather than a multitudinous

Figure 9. Arlon Ober at the Lambert Studio

maze of possible notes. The "?????" can be filled in with such descriptive words as Romance, Low-Slow Danger, Flying, Coldness, Heat, Open-ness, Military Might, Delicate Motion, Huge Build-Up, Swirling, Tongue-In-Cheek Brawl or virtually anything else that might kick off a visceral reaction in the composer. Chances are usually excellent that if the right word is conceived, the cue will be effective and appropriate. With the exception of sequenced material (i.e. the Tangerine Dream score to *Thief*, or the Giorgio Moroder score to *Midnight Express*), the basic contemporary scoring techniques for orchestra are still applicable to synthesized scores.

COMPUTER SEQUENCING

I do most of my synthesizer work at the Lambert Studio in North Hollywood, the home of the well-known Music Fantastic Library. Jerry Lambert and Will Roebuck have outfitted a room about 12x8 feet with a complete analog-digital synth and sampler set-up controlled by a Mac-Plus supplemented with a 40-megabyte hard-disk drive and running the OpCode "Cue" program and the MIDI information and video-sequence-synth sync is usually done through SMPTE code. (The next generation of all the music programs will allow transfer of MIDI files so that all pertinent set-up information such as beats, bars, tempo changes, etc. will not have to be re-entered, as is now necessary.) The wonderful thing about this room (*Figure 5*) is the extraordinary number of synth voices and samplers, which allow real-time performances of your work without the need to go to tape. Jerry goes to multi-track tape only when the amount of MIDI information is so great that it actually begins to affect the performance, or when doubling up of textural sounds—like a huge string section—uses up a number of samplers. Tape machines include a Sony (3M) 24-track and 4-track, a Tascam pro 8-track, and several 2-track mix-down machines. Outboard gear includes a variety of reverbs and processors including a Lexicon PCM-70 reverb, a Yamaha REV-5, multiple harmonizers, equalizers, and effects boxes. Synthesizers include a 4-bank Kurzweil rack-mount, a Kawai keyboard controller, a Prophet 5, a Jupiter 6 keyboard, Mirage keyboard sampler, and a Casio FZ-1 keyboard sampler. Rack modules and samplers include 3 Oberheim DPX samplers, an AKAI 5900 sampler, a Mirage rack-mount sampler, a Yamaha 416 module, several Yamaha 81-Z modules, a Korg DW8000 synth module, a Kawai K5-M, and a Roland D550 sound module. Though the principal sequencer program is the "Performer," a secondary system using the "Auricle" program on a Commodore 64 computer is also available to control unique tempo situations, as is also a Garfield Masterbeat interface. Drum machines include a Korg DDD-1, a Roland TR707, and Emulator and Kurzweil samplers.

Once the music is sketched you have two choices: fill it out on paper, or wing the orchestration on the computer or sequencer. Depending on the amount of time I have to input cue information, I may orchestrate all cues from the sketch page onto full score paper, or I may just begin to input from the sketch. If you input from the sketch alone, it is important to block out sections, chords, lines, and special effects with a notation of instrumentation.

The wonderful thing about inputting on the Mac-Performer system is the ability to edit voicings and to assign one voice to any instrument at any point. Some of the larger, all-encompassing computerized synthesizers, like the Fairlight, lock your input information onto a specific sound, and thereby making doubling, octave, or instrument changes impossible without inputting the voice anew. Even if you are unsure of where to go with any given cue's orchestration, once the basic individual lines are assigned to sounds and the drum machine, you can listen back to a real-time rendition of the entire cue and decide what areas need to be fleshed out. The sections of various voices may be bulk copied onto new ones and fine-tuned by adjusting octave and performance. This way, seldom is it necessary to input the same voice more than once, thereby saving a great deal of time. A wonderful shortcut to orchestration using the Performer program, is its ability to assign one part to multiple MIDI channels.

THE "STARTUP" FILE

At the outset of the scoring process, a "STARTUP" template is created. The "STARTUP" template includes *all* potential orchestral voices chosen for the entire film score, each with a specific MIDI channel assignment from 1 through 16. Each voice is assigned to specific synthesizers or samplers at the outset of the first working session and only need to be changed if an additional voice needs to be added to the list. In cases where there may be more than one part for an instrument (usually strings and horns), several parts are included on the template with common MIDI channel assignments. In most cases, 16 MIDI channel tracks are usually enough to accommodate all the needs for most of the films you may encounter. A sample template page is included so the layout can be understood (*Figure 6*).

When we start a cue, the "STARTUP" template is brought up on the screen, and the name "STARTUP" is immediately changed to include the cue reference name and number. Probably an easy way to think of this setup is as a telephone switchboard: dialing the correct number on the lobby phone gets you through to the various rooms. Now, if the violin track (MIDI channel 13) might sound interesting doubled by flute (MIDI channel 7), you can assign the violin to *both* MIDI 13 and MIDI 7 at the same time. This way the violin voice plays the violin sampler *and* the flute sampler without any additional inputting or editing. If you decide that only a portion of the violin track will sound good doubled with flute, you can reassign the voice back to MIDI 13 alone and copy the section that works onto a new flute track (again, MIDI channel 7).

Figure 10. Arlon Ober at his home studio.

If the percussion is being produced on a drum machine, the individual drum parts are programmed in from a keyboard—to allow all the parameters of touch sensitivity and aftertouch to be usable—and *all* assigned to the *same* MIDI channel. Most drum machines allow you to assign numbers to the individual pads that coincide with specific keys on the keyboard so that only those specific notes will trigger that particular percussion voice. For example, in this particular setup, the kick drum is triggered by the note F_1, the deep snare drum by E_2, closed high hat by A_2, open hat by C_3, etc. (*Figure 7*). This way, although all of the information is going into the drum machine, separation is maintained by the computer and drum machine recognizing pad triggers only from specific notes.

Once all voices written are inputted, additional material can be entered. The easiest and safest way to enter additional information for any one instrumental group (flute, horns, or percussion, for example), is simply to add a new track with the correct MIDI channel assignment and record from the chosen section. This will maintain the integrity of the original part and not risk erasing any prior information. If you feel the distinct need to insert the information in the original voice track, you can merge the additional one onto the original or you can assign the *record mode* to punch in and out at exact bars and beats. The record panel on most sequencer programs looks and functions like a real multichannel tape machine without the hassle of rewinding and with the addition of safety factors such as "UNDO RECORD" for when you screw up. ("UNDO RECORD" is my personal favorite.) The Performer also has the ability to overdub onto a particular part. However, I prefer to record overdubbed material separately so that I can isolate the new material. Specific mistakes in pitch, duration, multiple wrong notes, and volume may be isolated and corrected on the edit page of each instrument. Changing sounds of a particular voice may be done while the program is in PLAYBACK mode to hear how the combination will work. Once the preferred sound is found on the sampler or synthesizer, setting it for performance requires no additional work.

One extraordinary facet of this, and many other sequencer programs, is the simplicity of manipulating blocks of information either vertically or horizontally. We have the ability to capture a section of a single voice or a group of voices and gluing them onto new voices in the same time period in the same pitches, or transposed up or down (vertical), or to place them in a different time period (horizontal). This allows you to copy the base part from bars 1 through 4 onto itself from bars 5 through 9 and then copying the entire part onto the flute, 4 octaves higher. This makes structured pieces like songs and repeated patterns simple and quick to create.

QUANTIZATION VS. HUMANIZATION

Virtually all of the sequence programs for all computers have a "Quantize" mode which will smooth out any section of a performance, or the entire piece into exact, mathematically precise units which hit ex-

actly on the beat or specific beat increments. I'm sure most of you readers who have used any kind of sequencer program are familiar with this concept and most of you probably utilize this mode often. When I began using sequencers, "Quantizing" was the single most wonderful aspect of them. *The problem with them is that it de-humanizes the performance.* For futuristic, mechanical or mathematical effects, it's terrific. For rock, new age, and even Bach it may work most of the time, but for many other performances, complete quantization of a piece will effectively cancel the power and reality of an orchestral-style performance. *In fact, the beauty of human beings performing together is in the miniscule differences in their attacks, sustains, volume and phrasing.* The most effective way to enter most voices is to play each one in a comfortable tempo and only quantize those parts which normally require the most precision (i.e. drums, bass and some keyboard parts). Parts which are highly precise and fast may be either step-timed in or quantized only where necessary.

SEPARATING CONTROL INFORMATION

Initially, one should enter each part by just noting on and off information (*Figure 8*) and leaving out special performance traits like pitch bends, filter sweeps and crescendi-dimuendi to be recorded on a separate CONTROL track assigned to the same MIDI channel. The more information stuffed into a track, the greater the chances that a MIDI jam may occur. Separation of note information from performance idiosyncrasies assures you easy access to editing both and isolating problems. The most common problem with CONTROL track information (*Figure 9*) is that there is often simply too much of it. For example, bending a bass note down by using the pitch bend wheel for one particular quarter note may generate several hundred pieces of information in the time of one quarter-note. Not only is this unnecessary, but the chances are pretty good that so much information may cause performance slowdowns and mis-firing of notes.

CHANGING TEMPO INSIDE THE CUE

If you have calculated a performance (from CUE Music Editorial program or other similar programs such as the Commodore AURICLE program) which includes tempo changes, they are easily entered through a "CHANGE TEMPO" mode which works from any bar and beat to any other bar and beat. Tempo information can be entered as a straight line (indicating the same tempo throughout the section) or a slanted line (indicating an accelerando or ritardandi) between these two points. Optionally, the entire sequencer program may be externally controlled by the use of a Dr. Click or Masterbeat machine or the Auricle program on the Commodore 64. If you are recording onto multitrack tape, then a SMPTE code must be striped onto the tape first and the SMPTE track used to trigger the tempo-controller. With most MIDI-computer programs, a SMPTE-to-MIDI interface is necessary, unless the program itself has such capability as does the SMPTETRACK program for the Atari ST by Hybrid Arts. When I record at home, I use the SMPTE code striped on a multitrack tape machine to trigger the SMPTETRACK program on the Atari which, in turn, triggers the PERFORMER on the Mac Plus which fires the synthesizers through the MIDI information.

COST COMPARISONS

It is probably obvious by now that a great amount of information may have to be inserted to complete the performance information for any particular cue. The more thorough and dramatic you wish to be, the longer it will take to input and fine-tune your information. Though the time it takes to compose the material itself is probably the same as what a live score would take, the inputting procedure takes anywhere from one to two days per 10 minutes of music, depending on the number and complexity of the parts.

Where the computerized-synth score shines is in eliminating a number of potentially costly and time-consuming steps. Such steps as extracting the parts for orchestra, contracting players and studios, overtime for large groups of players, renting equipment and instruments, cartage of instruments and having to hire a librarian just to deal with the piles of paper involved, are totally eliminated. The down side is that the composer is usually safer if he oversees or does the note-taking and inputting personally, both highly time-consuming.

Tape costs are also diminished because many pieces can be mixed down directly onto 2- or 4-channel stereo (with SMPTE and/or sync-tone), avoiding the need to go multitrack. As I mentioned, the only time it has been necessary to go to multitrack has been when the amount of MIDI information is so considerable so as to slow down the performance, or in those cases that a number of synthesizers need to be employed to create a large section sound, as is sometimes the case with strings.

AESTHETIC DIFFERENCES

There are a number of reasons to argue back and forth for synthesizer or orchestral scores. The synthesizer composer often is burdened with numerous tasks not assigned to orchestral composers such as note-taking on the computer and inputting each voice. Obviously, a good keyboard player could assist here, and a programmer could help even more. Time needs to be taken to listen to numerous—sometimes thousands—of sounds prior to picking out a satisfactory orchestration template. However, much time and money is saved in orchestration, rental fees, and 2-inch tape costs. Synthesizers can produce some rather unique electronic and acoustic-type sounds and can reproduce most instruments now to the point where they are indistinguishable from the original. Saxes, guitars, and vocals still should be done live. But live orchestras also have a considerable up-side.

Live scores have the unique quality of richness and clarity that often becomes a problem with textured synthesizer music. Overtones tend to be additive in live recordings and often subtractive in synthesized scores. Dynamics and phrasing are easier and faster to get with people and machines can't provide the wonderful feedback that players add: their own personal additions of performance skills, taste and emotional power. There is an energy emitted by each and every orchestra with whom I've ever worked that makes one high and there is no experience quite like a live performance in a fine studio. But times and costs are asking for something new...a new, cheaper technology which can hold its own against the old methods, and computer-sequenced-synthesizers seem to fill the bill.

Every composer has a right and an obligation to express his personal opinions about the appropriate music for every film he works on. He should, though, take into consideration the cinematic needs, the producer's and director's wishes and desires and the music budget. Perhaps the most important thing he could do is assess the executive attitude toward the possibilities and proceed from there. If they make it clear that they hate synthesizer sounds, perhaps digital samplers are not entirely out. In fact, utilizing both real and synthetic sounds is often a practical and powerful alternative to many situations. Recently, a producer I worked for was about to settle for a small orchestral score performed in Hungary under rather uncomfortable circumstances. He was amazed to find what could be done locally with a digital-synth studio and ecstatic with the results. The new technology strikes again! I think even old Rip Schwartz might approve.

SUMMARY

Today's composer wears many hats. At times, he must be a music editor, composer, programmer, orchestrator, engineer, mixer, and music producer. He must know music, film, and the new technology.

He should know the classics, counterpoint, and the spectrum of the "pop" world. From this base of knowledge will his decisions be drawn, and the larger the base, the greater the possibilities. This doesn't mean to say that the entire effort can't be thrown out, but it does decrease the chance of this happening. Awareness of pop trends also allows one to develop at least some commercial instrumental or vocal material for every picture which may stand on its own as a record or song release and serve to increase public awareness and interest in the film itself. Today's film composer owes it to himself to become aware of the great potential of synthesizers and samplers especially in the light of changes music and financial trends in the film industry. Versatility is the watchword today. db

SANJEEV KUMAR

Sound Recording in India

This article is of particular interest in that it points out differences in approach dictated by cultural demands as well as the need to often make-do with existing equipment.

It was in the early forties that sound recording studios first appeared in India. Their primary purpose was to meet the requirements of the film industry and the commercial record companies.

The early studios in India needed only moderate facilities because of the simplicity of Indian music. Unlike harmony-based Western music—which requires a large number of participants in the orchestra—Indian music, being melody-based, has only a few artists. Also, the dynamic range of Indian music is very limited.

The introduction of harmony-based orchestral music in Indian films gave rise to the need for better quality multi-track equipment. However, the real technological upgrading of Indian studios started only after the establishment of Prasad Studios.

PRASAD 70 MM

Prasad Studio Complex is located in Vadapalani, Madras, India. Situated in the heart of the film industry, the Complex comprises of sound studios, dubbing studios, laboratories and many facilities needed by the film industry. Prasad 70 MM is the best and most modern set-up in the complex. Tastefully designed buildings, well laid out lawns and flowering gardens give the complex an elegant appearance.

Prasad 70 MM has many "firsts" to its credit:

- First studio in India to introduce non-perforated multi-tracks.
- First studio in India to start a rock and roll facility for mixing and ADR Suites.
- First studio in India to provide for 70 mm cinema production work.

The studio was constructed in 1978 and was in operation in 1979. It was designed to meet the needs of the 70 mm film industry. Due to the limited number of 70 mm films produced in India, the studio also fulfilled the requirements of the music recording industry by adding a 24-track recorder.

The technical staff are experienced sound engineers, graduated from the Film and Television Institute of India. Their assistants are qualified young trainees, trained to meet the ever-increasing demand for technical engineers. These engineers successfully service and repair the equipment as well as modify original equipment to meet local requirements.

The studio also provides facilities such as composing rooms, conference rooms and a few restrooms. There are no dormitories for overnight stay.

THE CONTROL ROOM

As per Indian standards, the control room is very spacious—approximately 40 M^2. The floor is raised up two feet by wooden planks and is thickly carpeted.

The walls consist of wooden compartments containing glass wool covered with decorative thick fabric. The rear wall is wood panelled and covered with thick, closed curtains.

Figure 1. The Prasad 70 mm Recording Theatre.

Sanjeev Kumar is an independent recording engineer in Bombay, India.

The acoustics are more towards the dead end—the rear—of the room. The front end, consisting of glass windows, is designed for live.

According to sound engineer Selvaraj, there were problems with LF hangover. This was minimized by the addition of panels and drapery. He admits he could not rectify the results completely, but the room sounds good without the use of an EQ.

The control room has only the mixing console and the monitoring speakers. The mixing console, consisting of an MCI LM 528 18/18 fitted with automation, was installed facing the massive glass panel to the front. A pair of Auratone 5C cubes were mounted on the console above the meter panel.

A host of ancillary equipment—such as AKG BX25E Spring Reverbs, MXR flangers, AKG TDU 6000 Delays, an URSA MAJOR Space Station, two KEPEX 4-channel noise gates, UREI/LA/44 channel compressor limiters and Roland DEP 5 effect processors—is fixed near the console for easy access.

The Tascam 52 ¼-in. mastering machine is brought in only when required to do the final stereo mixdown. The recordings are done on an MCI 24-track recorder which is kept in the machine room.

The main monitors—a pair of UREI 803s—are placed on a wooden platform at ear level, angled towards the console drawing on a matched UREI power amplifier fed via a Klark-Teknik room EQ always kept at bypass mode.

In spite of the well-known advantages of flush mounting the monitoring speakers, I have yet to see a studio in India do this. ("I have not come across the system," said the sound engineer. "I cannot give an opinion of it.")

Also, I have observed that in India the design of the control rooms is far from ideal. Unfortunately, in this country control room design is not given the attention it deserves. In some studios it goes to such ridiculous extents that the sound engineer has to know how a recording *should sound* on the studio monitor when the actual tape recording sounds normal in the concert hall!

Of course an explanation would then be forthcoming from the engineer for not equalizing the monitor flat.

Figure 2. The control room. Engineers Selvaraj and Bose are at the console.

The producers in India are used to listening to the 'high end' of the program, and if the monitoring is made flat, they feel that the recording lacks 'punch' and 'brightness.' So they leave it as it is.

THE MACHINE ROOM

The machine room is placed to the left of the control room with visual contact through a glass panel. The equipment is Westrex and perfect—one 35 mm film drive equipment, an MCI JH 16/24 track audio recorder, a Dolby rack and a Tascam 52 mastering machine. Other facilities include the Tascam 224, Technics double cassette deck, Westrex 6-channel mixing console, etc.

STUDIO FLOOR

The main studio floor is vast (approximately 550 sq/m and primarily designed and treated for a 70 mm mixing facility. It is also used for music scoring.

Figure 3. The machine room. Two Westrex film drive units being threaded.

The acoustic measurements were made similar to a 70 mm theater which has a brighter end. The RT is adjusted to three seconds and takes care of theater-type decay characteristics.

The entire floor is covered with PVC sheets. The ceiling angles down to avoid standing wave patterns. Up to 15 feet from the floor level, the wall is treated with 3-foot, wedge-shaped diffusers. Above them the wall is treated with an acoustic particle board. Glass wool is used under it in abundance. On the right of the studio floor are four spacious rooms for multi-track separations. These room are treated to achieve a very low RT. The voice-room beside the passage to the studio is spacious (Approximately 20 sq/m), treated with carpets and curtains. This room was made purposely brighter to give a little body to the voice.

The rhythm room on the left corner of the studio floor is again spacious and is acoustically bright. This room can accommodate up to eight musicians. The main studio floor can comfortably accommodate more than sixty musicians at one time. As the 70 mm work is limited, most of the work done here is film scoring.

On the far end of the studio floor, we have the large screen to project the 70 mm and 35 mm films for mixing and scoring.

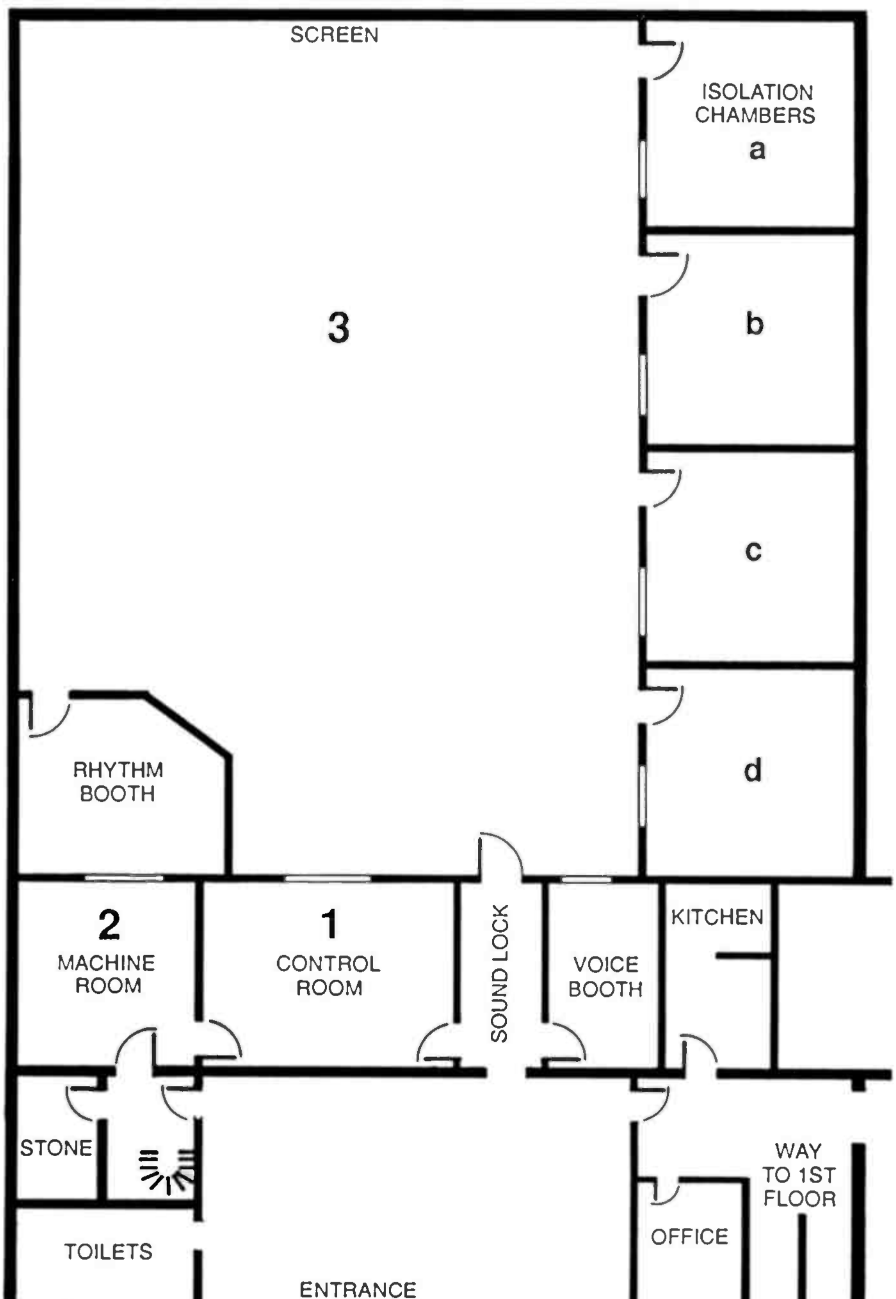
Owing to the disposition of the building, the projection room looks out onto the main studio floor, and the screen has been installed so that both can be used together.

Five massive Westrex monitors are placed behind the screen with 100-watts RMS for the 6-track 70 mm sound track. The effect track monitors are placed on both side walls.

In the main studio floor placed centrally is the Westrex-Litton ST 3070 M, fitted with additional panpots for the 70 mm mixing.

When asked about the microphones, Mr. Kishore said, "We all know that Neumann microphones rule the market. We have lots of Neumann microphones with us, the U87, U47, KM86, KM88,and KM84, and we also have an AKG 224, 451, 222, CKS 451 and a Sennheiser MD—441. We use condenser microphones for most of our recordings. They have excellent quality, especially for voice and strings. The strings," he went on, "we keep ten to fifteen of them on the main studio floor, near the screen."

Figure 4. Floor plan of the studio. A room over the front of the area contains the projectors.



TRANSFER AND PROJECTION ROOMS

Just beside the main lobby there is a staircase. It leads up to the room where optical and 70 mm transfers are done. The Westex 9000 series optical recorders, one for 35 mm and the other for 16 mm, and optical transfer for the Westrex 70 mm release print recorder are placed in this room.

Adjacent to this room, straight above the control room, is the projection room. It contains three projectors, two of which are 16/35 mm of reversible projectors (Westrex made) and the third is the 70 mm Westrex projector. The projection room is as spacious as the control room and is properly air conditioned to maintain a room temperature of 15 to 18 degrees C. The projectors are locally made by Cinecita under license from Westrex, Italy.

"The projection equipment has not given us any problems and it's running very stable," said the projectionist. On one side of the projection room are the Cinecita (Westrex) power amplifiers, six of them in a rack to supply the 6-track 70 mm. Adjacent to the projection room is an area where damaged films are repaired.

MAINTENANCE

According to our resident engineer Mr. Kishore, maintenance is not a problem. "Most of the problems are solved without too much waste of studio time," he said. "The studio is busy from nine in the morning until midnight, so preventive maintenance cannot be made regularly. We keep our work areas free from dust. They are air conditioned. Though our maintenance is not very cyclic, we are geared up. Once we see that the equipment is free from problems, we open up and check all parameters, thus keeping away all major breakdowns that could happen. We have had only a few major breakdowns. To me, a major breakdown is when we cannot locate the fault."

Prasad Complex is completely geared up with required maintenance equipment. Kishore is well-exposed and trained to look into equipment like Magnatech, Perfectone, Westrex, MCI and many others. He is considered one of the best engineers in India.

"When we cannot actually locate the problem, we take the advice of our manufacturers," he said. "But most of the serious breakdowns are attended by us. Spare parts are a big problem we are facing with MCI and Westrex equipment. Ever since Sony took over MCI they have not responded to any of our requirements. To talk about Westrex," Kishore continued, "the Mitsubishi Pro Audio at USA has clearly replied that they do not have anything to do with Westrex- Litton and that they do not have any old stock. What do we do with this costly equipment we have already purchased? Most of the time we are managing with locally-made components, but we are always on the look-out for original parts."

To prevent breakdown due to non-availability of spare parts, engineer Kishore has done certain modifications in the circuit with local components.

Certain areas of equipment tend to fail frequently due to the high temperature and humidity in India. "So we modify the circuits with locally available components to reduce breakdowns," explained Kishore. When asked about the alteration of specifications when altering circuits, he said, "We study the circuit thoroughly and then make necessary modifications. Technical specs would remain the same or turn out better."

Figure 5. The machine room. Note the MCI (now Sony) 24-track, 2-inch recorder and the Westrex/Perfectone film drive system.

COMPETITION

"We do not have any competitors," said Bose. "Most producers have their own studios. Therefore all the studios are busy. Then where is the question of competition?"

"Most people prefer having their recordings done here," said Kishore. "Only if we are booked do they go somewhere else."

Whatever the fact, the Prasad Complex is the best in India, and with their moderate charge, they have virtually wiped out any competition. Some producers come all the way from Bombay to have their songs recorded at Prasad. "As we have a 24-

Figure 6. The transfer room is located above the machine room (next to the projection room shown in Figure 7). A trainee engineer is operating the Westrex 70 mm release print recorder.

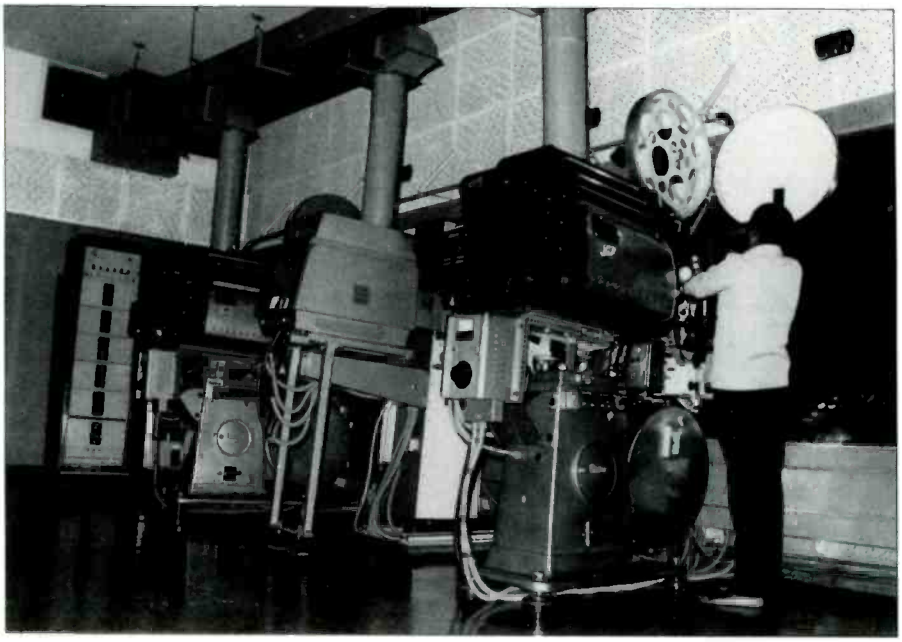


Figure 7. The projection room is located directly above the control room which has a screen at the far end.

track machine we can do our recordings much quicker than in Bombay," said Bose. "At 9:30 am musicians assemble. Music orchestration, lyric preparation composing and rehearsal is done before 1 pm. The singer comes in at 2 pm. Recording will be done and will be ready for transfer. After that we have an hour break. The second song starts and will finish by 11:30 pm. In Bombay it takes one full day to record a song. We are much faster than them. We have 40 to 50 musicians."

"We have also automation, which makes us work much faster," said Kishore.

MONITORING LEVELS

"We monitor at very high levels," said Kishore. "I mean alarmingly high levels. We engineers prefer low-level monitoring. When the session begins, the level would touch 90 to 100 dB SPL but the producers and music director would like higher levels and thus you find that as the session builds up, the volume also builds up. It is bad for our ears and bad for our monitors.

"Our previous monitor which was Altec was shifted to the ADR suite, due to its lower power handling capacity. The UREI 803s are excellent monitors."

WESTERN RECORDING COMPARED TO INDIAN

In Western music, the arrangement is excellent. In India, all the instruments play at the same time. We have at least ten different types of rhythm instruments, whereas in Western pop, you have only a few.

Our balancing is much tougher. The quality of Western recording is superb since they possess high grade equipment. However, I feel their balancing is not absolutely correct. The voice is often drowned in the music and is often inaudible. I have many times tried to follow the words and it is difficult!

Sometimes I wonder why the level of the singer is backed up. I should say that this happens in today's recording. But take the recording of the old days. The songs were intelligible and the balancing was good.

They use a lot of compression and distortion in the West today. I don't mean all the music is like that; there are many soft numbers with excellent orchestration and mixing. On the other hand, in Indian music, the lyric is most important. The audience expects to understand the *meaning* of the song.

EQUIPMENT

MCI LM 528 18/18 Mixing console with Automation

MCI JM 16/24 24-channel multi-track 2-in. recorder

Final mixing console 70 mm (mixing) Westrex-Litton ST 3070 M with additional panpots for 6-track stereo

Westrex 6000 series (five units) film drive equipment 6-track facility with four noise reduction systems

Perfectone film drive recorder 4-track Copermag

70 mm Westrex release print recorder

Westrex 9000 Series 16 mm and 35 mm optical recorder

Westrex-Litton 8-channel input mixing console

FP 30 Kinotone 35 mm projector

35 mm / 70 mm Cinecita/Westrex projectors

UREI 150 watts/channel power amplifiers (Control room)

Auratone Cubes 5C

UREI 803 monitor speakers (Control room)

Westrex/Cinecita 100 watts mono amplifiers for 6-track 70 mm

AKG BX 25 E Spring Reverbs

OUTBOARD EQUIPMENT

MXR Flanger

AKG TDU 6000 Delay

Ursa Major Space Station

Kepex 2 Noise Gates (4-channel Plug-in type

UREI LA-4 compressor/limiter (4-channels)

Philips line mixer 6 inputs

Klark-Teknik 16-band stereo graphic equalizer

Neumann microphones—U87, U47, KM86, KM88, KM84, AKG 224, 222, 415, CK5, 451 with preamp, Sennheiser MD 441

Tascam 522 mastering recorder

Teac 224 4-channel cassette recorder/mixer

Technics Stereo cassette

ED LEARNED

Sound Reinforcement in North Africa, Part II

MOROCCO

Morocco was an important part of any North African tour: because it is the closest country in the region to the U.S., it marked either the start or finish of a trip. The Jay Hoggard Quintet tour of 1985 began in Morocco. We flew from New York to Paris' Orly Airport, where most of the flights to Africa originate. Unfortunately, fog prevented our landing, so we were diverted to Zurich, Switzerland. We returned later that same day after the weather cleared, landing far too late to make our connection to Morocco. Early morning fog at Orly is common, so if scheduling allows, afternoon connections are preferable. We finally arrived in Casablanca around 9:00 PM, a good five hours late. Customs clearance for the group was a snap: unlike most of North Africa, no visas were required, and the USIS-Morocco staff did a fine job of easing us rapidly through customs. We left the airport for Rabat, about an hour's ride by car.

AUTHOR'S NOTE:
In this issue's edition of my North African sound reinforcement notebook, we'll look at each country individually, concentrating on concert venues and performances. My three tours of the region spanned the last four years; changing politics in the region occasionally affected transportation into and around certain countries. The challenges posed by these changes, and our solutions to specific problems, will be covered on a per-tour basis. The sound system used at each concert was the stock USIA PA described last issue unless otherwise noted.

Rabat

Although our concert was scheduled for Saturday, we arranged to get into the venue on Friday to set up and rehearse. The Mohammed V National Theater was a very modern facility, with a capacity of 2,000, including a very large balcony. Voltage was 240, and power could be obtained offstage directly from ungrounded European-style receptacles or directly from a drop via terminal posts.

The local police were on the lookout for these electrical thieves; I had visions of ending up in a Moroccan jail...

An equipment ground was provided at the drop, so I elected to tie in there and pass on the receptacles. Once we were set up, the group played for around three hours, both performing songs from the concert set and working on some new material. The hall's acoustics were excellent: carpeting and other treatment made it just live enough without being overly reverberant. I had a chance to refine my mixing of Jay's quintet *and* check out the PA. The hall was too large for the Bose system alone, so I also used the house PA system for augmentation. The house PA was comprised of eight hi-fi speakers, located along the proscenium opening. I quickly discovered that any serious low-frequency information caused these cabinets to distort, so I only used them for acoustic piano; the instrument that needed the most help. We had bassist Jerome Harris give us more low-end from his stage amp; the guys on stage could still deal with it, and I didn't have to count on my small system to push as much bass.

Casablanca

Our concert here was held at the Municipal Theater, a small venue seating 600, although capacity had been increased to 700 via folding chairs on the sides and in the back. The seating was all on a single level, raked slightly upwards from front to back. The audience area was not carpeted, and the plethora of hard surfaces yielded a reverb time of almost two seconds. Power was my biggest problem, however. This facility had a voltage of 220, comprised of double 110s; each receptacle had two 110-volt hots and nothing else. I asked the house electrician to give me a single 110-volt line and the neutral from his power board; as ground was non-existent, I elected to use a water pipe that was a few feet from the power board. I discovered that my hot was really 130, and the neutral carried a whopping 35 volts! The house guy insisted the problem lay in the city system; I thought otherwise. With escort Ward Kirchwehm along to translate, the house tech and I raced from point to point, always finding the same bad neutral. Finally we did something very novel: he led me to the street, where we pried open a manhole cover in the sidewalk and climbed down into the main city electrical tunnel. Ward stood watch for us—the electrician told us that people often tapped into the line themselves, bootlegging AC without being metered.

The local police were on the lookout for these electrical thieves; I had visions of ending up in a Moroccan jail for insisting on a clean neutral! The tunnel was impressive; it contained huge bus bars and giant wires. We discovered, as I'd suspected all along, that the neutral was fine *before* it entered the building. We finally managed to locate a drop, located behind a dimmer rack offstage left, where the neutral carried an acceptable 2 volts. I borrowed a step-up transformer from the house techs to run my 220-volt Crest power amps. Everything else was run at a U.S. standard 120 volts, bypassing all internal step-down transformers.

Dealing with the live acoustics and small hall size called for some novel solutions. I had guitarist Vernon Reid and Jerome place their amps on the side of the stage, angling them across so the sound stayed on stage and didn't spill into the house. Everyone could play at the level they liked without overly exciting the room. The down side of this was that I had to crank the monitors higher to get the acoustic piano audible over the increased stage volume. We only had half a house that night, but they were overwhelmed by a torrid duel between Vernon's guitar and Jay's vibes on the bandleader's own *Mariposa*.

One note about the show: the concert was not only sold out, it was oversold! About five-thousand people attended...

A DIFFERENT VIEW

The Terrance Simien tour in 1988 gave me a different view of Morocco. We played exclusively in the western part of the country, and we no longer had a PA system traveling with us. The USIA system we'd used for the first part of our tour had been promised to another group, and was shipped back to Paris from Cairo. While I did carry my own microphones and a modicum of signal processing gear, we were totally dependent on local PA systems. The band had to play with locally procured amps; fortunately, USIS-Rabat owned a Peavey Bandit 65 guitar amp and a TKO bass amp identical to the ones we'd been using.

Figure 1. A view of the stage and mix point, Salle Couverte, Oujda.

We entered the country by car from Algeria; we'd flown from Algiers to Tlemcen, where U.S. consulate vehicles from Oran were waiting for us. They drove us to the Moroccan border, where we were met by vehicles from the U.S. embassy in Rabat. Despite the special releases we'd received from officials in Rabat, it took a good two hours to clear customs.

Part of the problem was that our gear came in by land, but would leave by air. To prevent customs holdups, arrange to have equipment arrive and depart by the same mode of transportation. Once through customs, we continued on to the border town of Oujda, where we began our western Moroccan concerts.

Oujda

Terrance's group was the first American band to ever play in Oujda, and the whole town was excited. Our concert was held in the "Salle Couverte" of the Ministry of Youth & Sports. In plain terms, it was a large gym that seated around 4,000 using folding chairs on the gym floor and bleachers around the sides (see *Figure 1*). Acoustics were horrid despite some impromptu floor covering—very bright sounding, with a reverb time of an unmanageable three-and-a-half seconds.

My local PA system was based on a Montarbo 24x4x2 console, with four monitor sends and two effect sends. It also had a stereo octave-band-width graphic on the main outputs. These graphics were not patchable, however, so while I had EQ for the mains, I had nothing I could use on the monitors which were two self-powered Montarbo floor monitors. My house system was a Montarbo component stack (one bass bin, one mid cabinet with two 12-inch woofers, and a high pack with a 1-inch driver/exponential horn combo and six piezo tweeters) and a Yamaha column speaker per side (see *Figure 2*). USIS-Morocco brought along a small step-down transformer for my 110-volt outboard gear. Power here was 220 volts, supplied on ungrounded European-style receptacles.

I patched one of my two SPX-90s into the monitor send, selecting the parametric EQ program. While not as effective as a ⅓-octave graphic, it was better than nothing; I was able to get just enough gain-before-feedback. The group helped the situation by cutting back on their stage volume. I used a similar approach in dealing with the house sound—less was definitely more in this extremely reverberant space.

One note about the show: the concert was not only sold out, it was oversold! About five-thousand people attended, filling every nook and cranny of the gym. Their response to the group was overwhelming; we had to stop the concert several times to clear the stage and apron area of overzealous fans. Fortunately for all concerned, the crowd

Figure 2. A closup of the Oujda P.A. stack.

was very well-behaved, and promptly responded to the pleas of concert organizers, although the music always incited more dancing. From an audience reaction standpoint, it was our best concert of the tour.

Fes

The drive from Oujda to Fes took about three-and-a-half hours. The following day's concert took place at the Complex "Al Qods." This was a rectangular room with an elevated permanent stage at one end, playing to the room's short dimension. It seated 600, in folding chairs placed on the tile floor. Power was available stage right on an ungrounded European-style receptacle; voltage was measured at 226. My PA was a Dynacord Eminant 1040M console, a 10 x 2 configuration, powering two Dynacord E152 speakers on tripods. Each speaker contained a single 15-inch woofer and a horn/driver combination. The monitors were two Dynacord MC50 cabinets, each containing a single 12-inch woofer with whizzer cone for extended highs. The back wall was fairly close to the stage, and with a reverb time of 2.5 seconds, I decided on minimal reinforcement; the drums and bass amp were not amplified. I also elected to use some of the sound gear that our USIS-Moroccan staffers had brought from their office. Two Peavey 1154HS floor monitors and a Peavey M-3000 mono amplifier were employed; these went to Terrance and guitarist Sherm Robertson, our two vocalists. The Dynacord wedges were assigned to cover the rest of the band. With ⅓-octave equalizers in short supply, I was again forced to use one of my SPX-90s as a monitor equalizer. Another capacity crowd enjoyed this concert, although things never did get as crazy as in Oujda.

The tricky part would prove to be getting the stuff out of Algeria by land after bringing it in by air.

Meknes

This city is situated about an hour by car from Fes. The concert here was held at Cinema ABC, a local movie theater with a small stage in front of the screen. It seated 1,000, including a small single balcony. Power was identical to Fes in voltage and receptacles. The hard walls and concrete floor contributed to a reverb time of two seconds.

Two different pairs of H/H cabinets were provided for house, per side: a Pro Series II which contained a 12-inch woofer and bullet tweeter, and a Dual Concentric which contained two 12-inch woofers, each with whizzer cones for extended high-end. A LEM Compact 210S 10-channel powered mixer was provided for mixing and PA power, but to my horror I discovered it had only high-impedance inputs, and I carried only low-impedance mics.

To top it all off, there was no monitor system. The local high-Z mics were of questionable quality, so some improvisation was called for.

We called Fes and arranged for the same sound guys to bring low-Z mic cables and their Dynacord wedges to Meknes. In the USIS sound equipment were: a 6-channel Peavey mixer, a 601R; and a 6-channel Peavey mixer/amp/octave graphic EQ, a 600B. By running the 601R into an aux return on the 600B, I effectively created a 12-channel mixer with low-Z inputs. I used the amp on the 600B to power the two Peavey wedges, and the amp on the LEM mixer to power the H/H PA cabinets. The Peavey M-3000 amp powered the wedges from Fes.

The on-board Peavey graphic EQ became my monitor EQ while the LEM graphics handled the house EQ. With the help of a few Y-cords and some DI Boxes as isolation transformers, I was able to get this patched together system to work without buzz or hum—much to the amazement of the local technicians who bombarded me with questions about interfacing sound gear. Although we only had half a house that night, several dignitaries were in attendance, including the Governor of Meknes, so my efforts to insure the best possible sound were appreciated by all.

ALGERIA

Terrance Simien's brief five day stop in Algeria preceded our tour of Eastern Morocco. We arrived in Algiers via an Air Algerie flight from Cairo, Egypt. The efficient USIS-Algiers staff had all the necessary documents to insure a smooth entry. The tricky part would prove to be getting the stuff out of Algeria by land after bringing it in by air. We gave three concerts in Algeria; all were in and around the city of Algiers.

Our first concert was held at the Salle Ibn Zeydoun, a modern theater located in a new shopping mall complex. It seated 600 in a steeply raked main floor area. The whole theater was thickly carpeted, and featured a lot of acoustical treatment on the walls and ceiling. This room was very dead; reverb time was less than a second, making it perfect for Terrance's amplified group. The house PA was built-in behind a false wall on either side of the stage. It was in stereo, and featured JBL components powered by UREI 6300 power amps. There were four JBL monitors, with a single 12-inch woofer and a bullet tweeter in each. UREI 1535 equalizers and 2562 feedback suppressors provided EQ; UREI LA-4s were used for system limiting. A portable TOA RX-216 console rounded out the package. Our band gear, rented from a local group, proved to be a Peavey 400 bass amp with a double 15-inch bottom, and a Yamaha GL 100 guitar amp. I borrowed a step-down transformer from the USIS office for my personal front-of-house gear, as power here was 220 volts.

Speaker placement was a bit problematical as I really had to cover two separate areas.

The usual mix location was a glassed-in booth at the rear of the hall but I persuaded the sound guys to set the console in the theater's rear aisle—immediately in front of the booth. Patch points were located on the wall behind me, so I could access snake lines and amp inputs easily. Total system power was around 2400 watts and the combination of real power, good PA, great acoustics, and a full house combined to give us the best sound of the tour. The crowd was whipped into a shouting, dancing frenzy by the group. I took full advantage of the stereo PA to indulge in some panning, including some dry-wet effects panning that had the house techs running over to see what I was doing as echos moved across the hall!

Figure 3. From left to right, Roy "Chubby" Carrier, Earl Salley, and Terrance Simien checking the acoustics at the DCM's house.

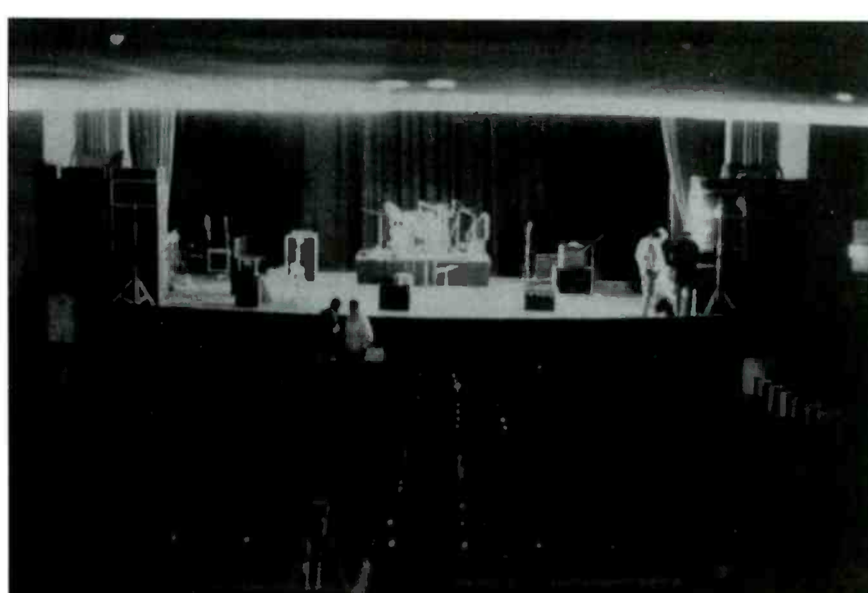
CHALLENGES

Variety of venue is one of the challenges we face on a typical government tour and our next concert required a complete change of philosophy.

We were to play an invitation-only concert at DCM (Deputy in Charge of Mission) Chuck Brayshaw's residence. The band would set up in a tiny room that opened up into both a larger room and a long hallway (see *Figure 3*.) We used the same band equipment and borrowed some mic stands, two JBL monitors, and an APK 2230 power amp from the Salle Ibn Zeydoun.

USIS provided a Peavey 600B 6-channel mixer/amp and two Peavey 1210HS PA cabinets. The amp in the Peavey 600B powered the mains while our borrowed APK powered the monitors. Speaker placement was a bit problematical as I really

Figure 4. The view from the mix point to the stage of the Salle Atlas.



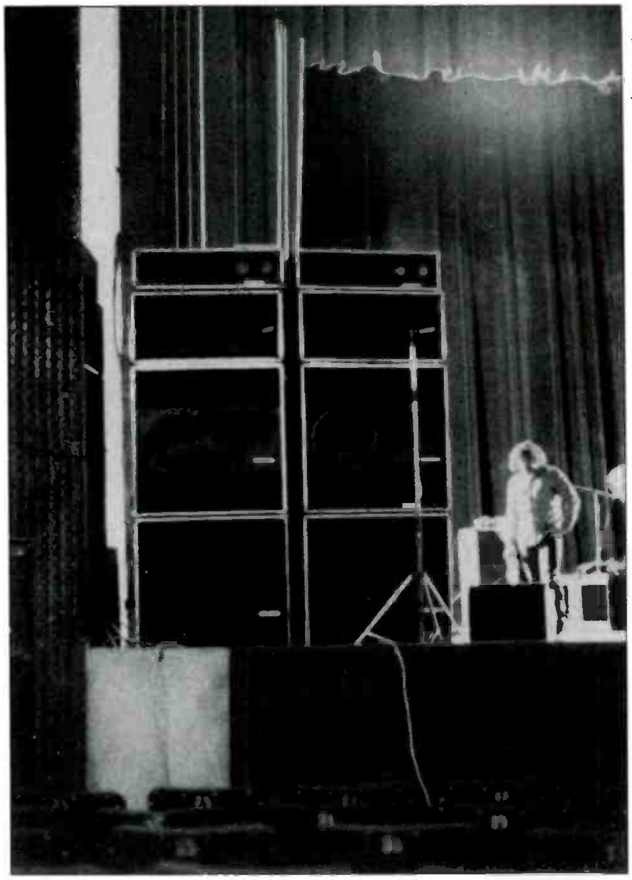


Figure 5. Speaker stacks at the Salle Atlas.

had to cover two separate areas. I placed one speaker, in the corner of our large room, on top of a table to elevate the cabinet to head height.

The other speaker was placed at the end of the long hall, firing down it's length to cover the hall and the area immediately in front of the band.

Considering the small room and the fact that I only had six channels available, I elected to amplify only the vocals and accordions. By carefully balancing the band's stage volume, we were able to get a sound that was representative without blowing people out of the house. My other little hassle, however, was with AC. Located in a U.S. diplomatic enclave, the power was 110 volts. The PA gear was all U.S. standard 110, but the band gear and monitor amp were 220 volt, so we had to scramble to find a step-up transformer to handle our Algerian gear.

Our final Algerian concert was held at the Salle Atlas, located in Bab Al Oud on the outskirts of Algiers. This proved to be a large theater, seating around 1,000 in a single floor area (see *Figure 4*). There was a balcony area, but it contained no seats due to an ongoing renovation. Acoustics were not pleasing: reverb time was around two-and-a-half seconds, and the wood and plaster surfaces contributed to a very bright overall sound. House PA equipment was substantial: per side, there were four Boyer bass bins with two 15-inch woofers each, two Boyer horn-loaded mids with two 12-inch woofers each, and two Boyer horns with a 120-degree horn/driver and two bullet tweeters each (see *Figure 5*).

I spent half my time listening to headphones to cue the T.V. mix, ripping them off to listen to the house mix, and then putting the "cans" back on—definitely hard on your hair!

Three LPS 5500 power amps were used for each side of the mains. Front-of-house gear included: a Soundcraft 400B 24-channel console, two Dynacord and Roland ⅓-octave graphic equalizers, a Nexo 4-way electronic crossover, a Roland SRE-555 chorus echo, and a Dynacord digital reverb. A Peavey Mark IV 16x8 monitor console and a rack of six ⅓-octave equalizers provided stage monitor control. There were two types of floor monitors: an APG which contained a 15-inch woofer, 8-inch woofer and a tweeter, and a Boyer which contained a 12-inch woofer and a 90-degree horn/driver. Both were passively crossed and powered by H/H S500D power amps.

I found that the APG 3-ways got louder than the Boyer monitors, so I assigned an APG to Terrance, Sherman and bassist Popp Esprite. Both Boyer wedges were assigned to drummer Chubby Carrier. The local sound crew provided a monitor engineer for us. With translation from escort Ron Minninger, the guys were able to communicate what they needed to hear. I had my hands full with the PA; the overall sound was horribly bright—lacking any real warmth.

The reasons for this were twofold: some of the woofers weren't working, and also the crossover settings were full-up on every section. By using a combination of crossover gain controls and amplifier gain controls, I re-balanced the PA for a smoother sound, much to the delight of the house guy who followed me around taking notes! Our concert was recorded by T.V. Algiers who required a special feed from me.

I arranged to give the T.V. a pre-fader aux output, giving me the ability to create a completely different mix for the taping. Because of the live nature of the Salle Atlas, I didn't mix things "up" enough for a PA mix to be adequate for recording purposes. This did create some juggling for me during the show—I spent half my time listening to headphones to cue the T.V. mix, ripping them off to listen to the house mix, and then putting the "cans" back on—definitely hard on your hair!

The band was very happy with the sound (I'd made a cassette tape of the T.V. feed) despite the fact that the monitors had not been up to par.

Apparently, the stage engineer had made all sorts of changes during the show to please himself, not the band. Despite our most complex monitor system of the tour, the band was eager to go back to me mixing monitors from the house.

Who would have believed it?! db

BRENT HARSHBARGER

Audio for the Church

• This issue we are going to discuss the device that more often than not is considered the "magic" device of audio engineering, the mix jockey's delight—the mixing console.

First, we are going to discuss what the console is and how it works, and in the next issue we will discuss real world "hands on" applications.

The mixing console is my personal favorite device in audio, because as a child I loved to push buttons and tweak knobs. I never grew out of it, in fact, it became a time-robbing infatuation.

What's your excuse?

CONSOLE OVERVIEW

Although the console is my, and most audio engineers' first love, it sometimes is the most overrated device in audio. Why would I say such a thing?

Well, if you take the so-called magic out of it, the mixing console is an audio routing device, nothing more. The console has inputs and outputs. Everything else just tells the sound where to go and how loud it can be. In our first illustration we are going to look at a block diagram of a basic channel on a mixing console (*Figure 1*). First, you have a mic plugged into a channel. Initially, mic level is very weak and has to be amplified; the mic pre-amp is the first stage in the console's signal path.

Next, we go through an equalizer circuit, then to the channel fader and output. That, in a nut shell, my friends, is the only thing the mixing console does (or at least in a simple form). As you spend more money for more knobs, you get more outputs, and you can select how much goes to what output.

LOOKING DEEPER

Now we are going to build on our basic concept of an input channel and look at how the signal flows through it. My second illustration shows us looking down the top of the mixer, so that you can see the controls that we will be working with (*Figure 2*). There are six rotary knobs, or pots (short for potentiometer)—the first being the gain, followed by the EQ section, monitor, effects, pan, and finally the slide fader.

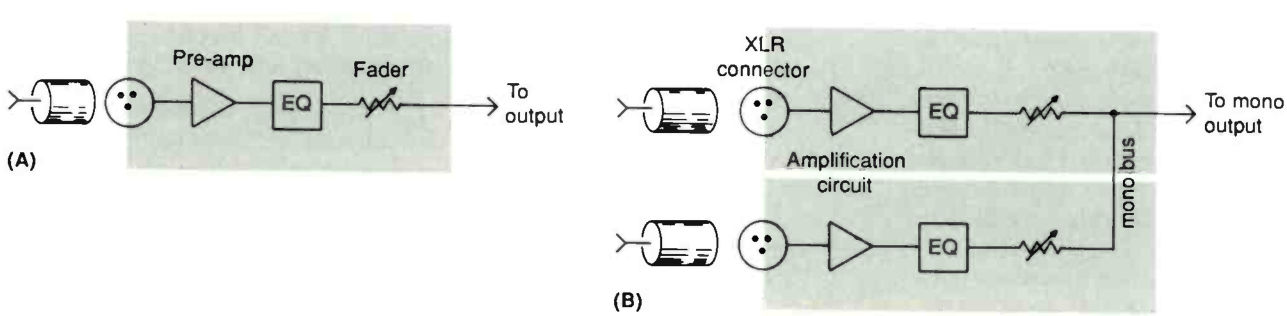
As the signal comes in the channel (*Figure 3*), it is a weak mic signal and has to be amplified. The gain pot control determines the amount of amplification that you will bring the gain up until you get the optimum level. (We will discuss this in detail in a later issue.) Next, it is time to go through the EQ circuit. The EQ circuit is where some of the so-called magic comes from. Keep in mind that I can do what we have talked about thus far by running a mic through a mic pre-amp box and coming out of that into a graphic equalizer. I'm sorry if I took the fun and mystery out of the console, but the console is not as magical as we sometimes think it is.

After all, the EQ circuit in the mixing console is just a routing device. However, here is where the console now gets a little more complicated—but not impossible, it just takes more thought about what is going where. How well you understand the logic behind what is going on here will determine how well you will be able to master the console. No matter how large or what make they are, they all use the same logical concept.

MOVING RIGHT ALONG

To keep everything simple, we will continue with our illustration (*Figure 4*) after the signal goes through the EQ circuit. There, it comes to a monitor pot which sends the signal down the monitor bus to a summing amplifier, which then goes through the monitor pot or fader, and out the monitor output jack. Now the thing I

Figure 1. A basic channel on a mixing console.



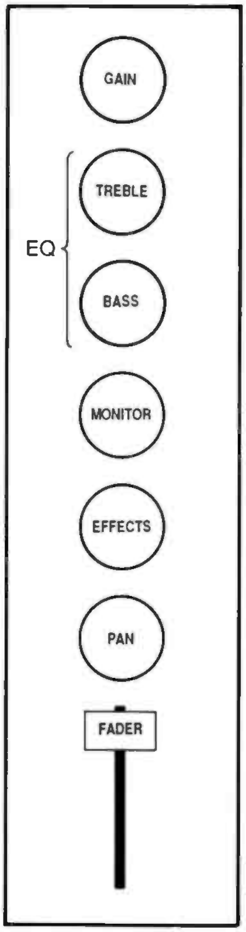


Figure 2. Looking down at the six rotary pots.

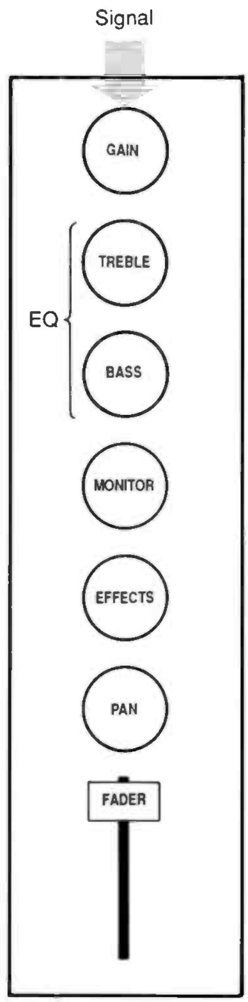


Figure 3. The gain control adjusts the level of weak mic signals.

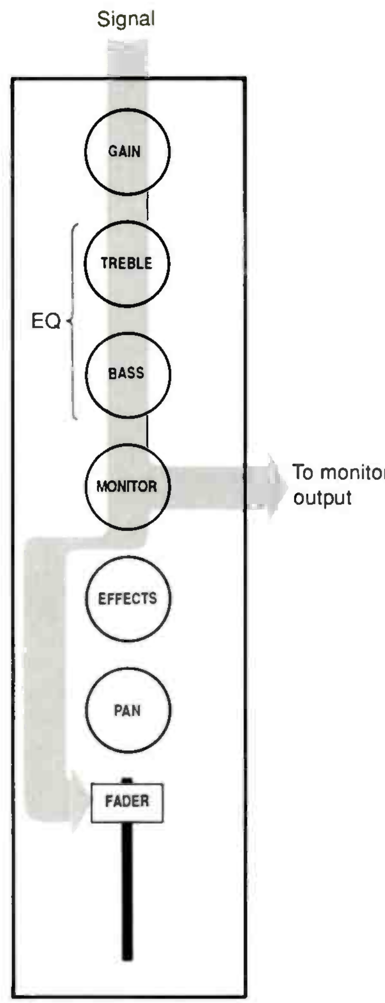


Figure 4. The monitor pot sends the signal down the monitor bus.

want you to notice is that the signal that comes out of the monitor output jack is a signal that is affected by the EQ circuit. Accordingly, any EQ adjustments made will also affect the monitor mix.

Each time the signal flows through a circuit, that circuit affects anything down line of that circuit. So far, we have talked about the monitor output, but the two things that the signal flows through before it reaches the monitor pot (and bus) are the gain and the EQ circuits.

Therefore, any adjustments to these circuits will be passed on down the line. The illustration in *Figure 4* shows us what we call a post-EQ, pre-fader configuration. The monitor signal is after the EQ circuit, but it comes before the fader, so any adjustments to the fader do not affect the monitor. Most monitor systems use a post-EQ/pre-fader, so that the musicians can have an independent mix in their monitors. This way, moving the faders up and down for the house needs doesn't change the mix to the musicians.

Now, we can take a look at the effects send which is a little tricky because it is a post EQ/post-fader configuration. Notice in *Figure 5* that the signal flows through the gain, EQ, then makes a U-turn through the fader, and then to the effects send. The fader now affects the effects control. Therefore, if the fader is closed (all the way down), no signal is sent to the effects bus. One reason for this is to keep the reverb or delay signal at a proportional level to the house mix.

The pan circuit (*Figure 6*) is similar to the effects bus because it is a post-EQ/post-fader control device, but is divided into two separate busses. The pan pot is a resistive network. Resistive networks (circuits) work by the current taking the path of the least resistance (or opposition), so if you turn the pan pot to the left, it puts heavy resistance on the right bus.

MORE FEATURES

Now, as in real life, we've outgrown our console, so it's time to move on to a board with more features (usually not as in real life). Most consoles, even the least expensive ones, have an insertion jack on each channel. How the jack works is: the mic signal comes in, goes through the gain circuit, and then to the EQ. However, if you have an insert plugged into the insert jack, it interrupts the signal after it goes thru the gain stage (refer to *Figure 7*) and returns before the EQ. Looking at the insertion jack and plug, see *Figure 8(A)* and *(B)*, you can see a little easier what is going on because the send and receive are on a single jack. The jack is a three conductor tip, ring, sleeve, and ¼-inch plug (commonly known as a stereo plug). The tip is the return and the ring is

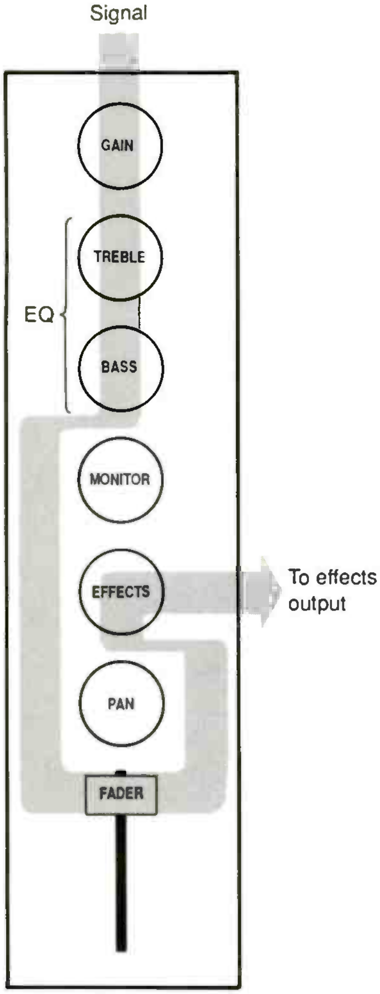


Figure 5. Signal flows through the gain and then U-turns onward.

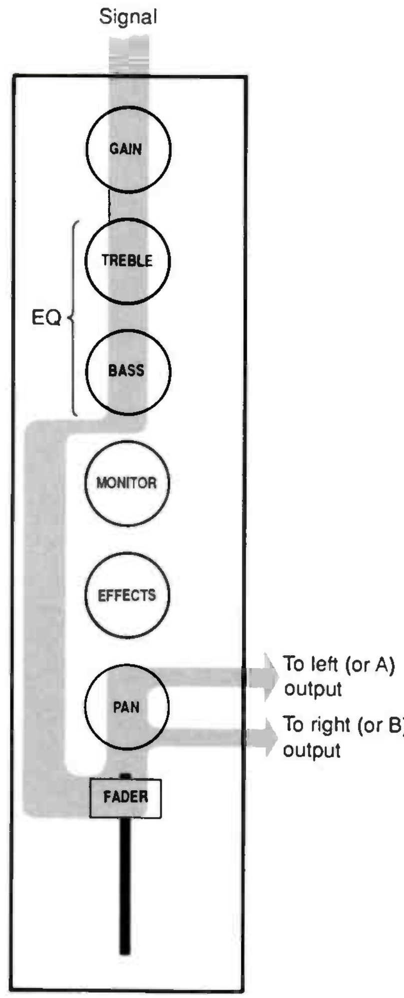


Figure 6. The pan circuit is similar to the effects bus.

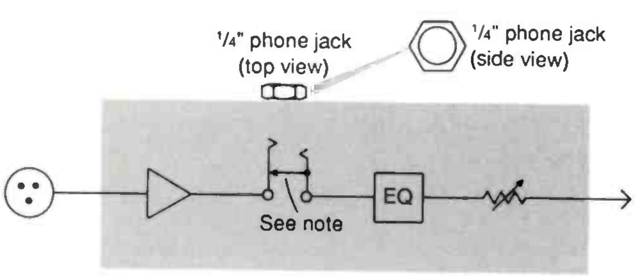


Figure 7. Note that this acts as a switch. If nothing is plugged in, signal flows through. If a jack is pushed all the way in, it breaks the contact.

the send which leaves the sleeve as a common. The shield of both cables is terminated to the sleeve prong of the plug.

The first thing I want you to notice about the jack is that if you insert a plug only as far as the first prong in the jack, it doesn't interrupt the signal path, but as soon as you plug it in all the way, the jack acts like a switch and interrupts the signal path. This is because the plug forces the second prong to break contact with the signal going out of the jack, and makes the outboard device a part of the channel's circuit.

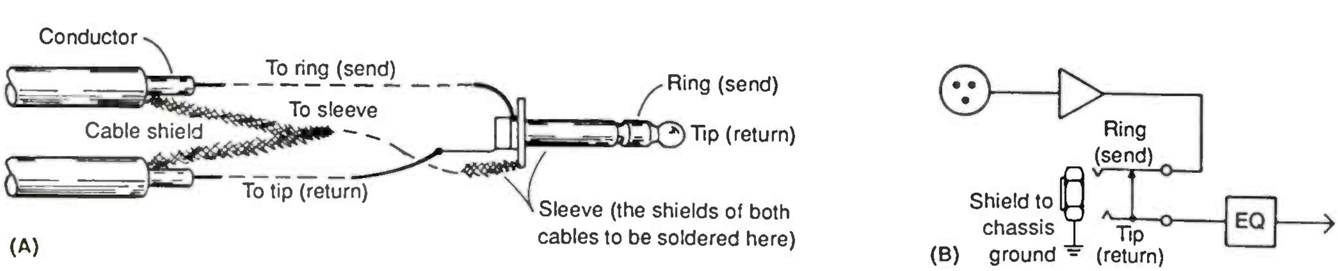
The second thing is, if you keep the plug at the first position, you now have a pre-EQ/pre-fader send which you can use to send a channel for channel from a PA console to a remote recording console. As long as the person operating the PA console doesn't change the gain pot, you have an independent mix at the recording console. But if they do change the gain, you quickly have to recover your recording mix.

SUB GROUPS

A helpful feature available on intermediate-to-higher-priced boards is a feature called a sub-group. Sub-groups make a large mix manageable. If you are mixing a large drum set, several keyboards, and many singers, it can be a nightmare to try to mix. By assigning the drums to group 1, keyboards to group 2, and vocals to group 3, mixing is made easier by "fine tuning" the mix with the three group faders instead of 16 to 32 channel faders.

A sub-group in it's simplest form is a stereo bus. For example, you can assign all the music to the left output and assign all the vocals to the right output, by turning the pan pot of the appropriate channel. Sub-groups are typically in groups of 4 or 8 on PA

Figure 8. At (A) you see the wiring configuration that is connected to the jack that is shown at (B).



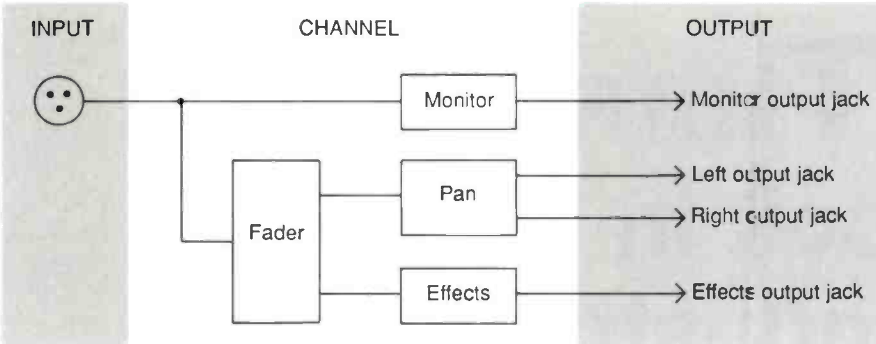


Figure 9. Signal flow in a console.

type consoles. The sub-group circuits are all the same in concept, but it is best if you refer to your owner's manual to insure that you are operating the console in the correct way in which it was designed by the manufacturer.

Subs (for short) work as I mentioned above, but use more than two outputs, such as the example of the stereo bus. Plus, on almost all consoles, each sub can be individually panned left or right into the stereo bus. Therefore, you are getting 16 (channels) X 4 (subs) X 2 (your stereo bus) or simply, 16 channels assignable into four sub-groups into a stereo bus.

Unless you attend a larger than average church, I've covered the features of what most of you have in your church. If not, or if you have any questions or comments, please write to me at **db Magazine** and if you're writing with a question regarding your mixing console, please specify make and model.

In closing, I would like to show you the last illustration (*Figure 9*) on the signal flow of a mixing console because it is vital for you to master the operation of mixing consoles. In the next issue we are going to test your "console" mental skills, and help you master all your Sunday mixing applications.

See you next issue! db

LEN FELDMAN

Lab Report

Fostex D-20 Digital (DAT) Master Recorder

GENERAL INFORMATION

While the consumer world waits for the much delayed introduction of consumer type digital audio tape (DAT) recorders in the U.S., the professional audio world has pounced upon the new R-DAT format enthusiastically. "Professional" DAT machines have found their way into most recording studios and broadcast facilities and are serving as mastering tools, often replacing or augmenting analog mastering decks as well as more expensive stationary-head digital tape recorders.

Most of the professional DAT recorders we have seen thus far lacked one important feature: the ability to record and read SMPTE Time Code. In their D-20, Fostex has taken care of that format omission (the R-DAT format was originally intended for consumer use), by altering the DAT format so that SMPTE/EBU time codes can

Figure 1 (A). Frequency response plotted from below 10 Hz to 20 kHz. The lower trace is right channel output. Results are plotted from a digitally-recorded test tape.

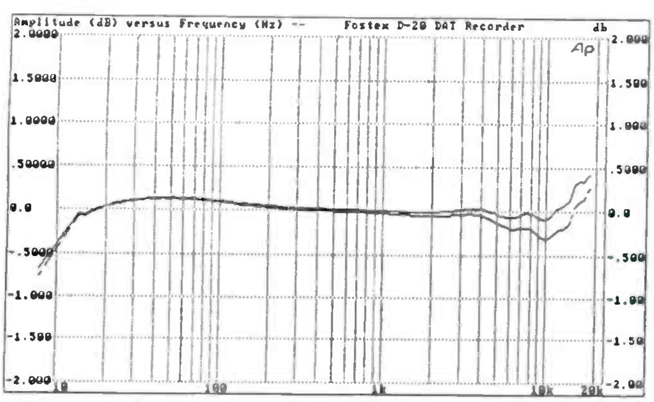
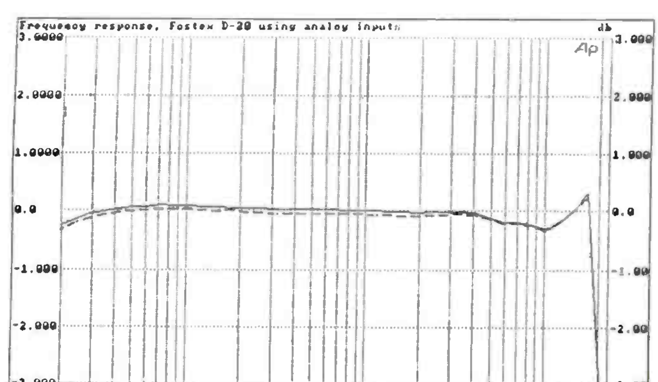


Figure 1 (B). Frequency response using the analog inputs.



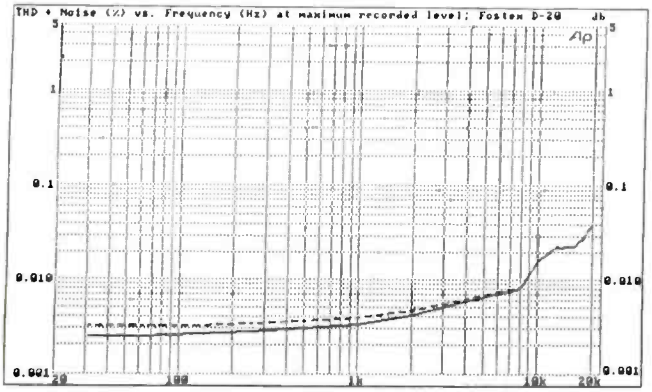


Figure 2. Harmonic distortion plus noise versus frequency, expressed in percent relative to maximum output level.

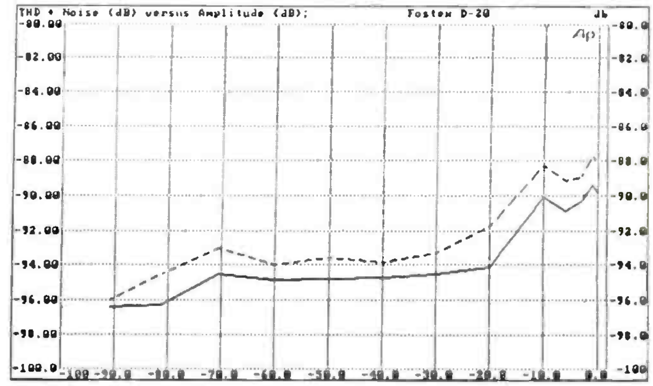


Figure 3 (A). THD plus noise (referred to 0 dB recording level) versus signal amplitude.

be recorded in the sub-code data area of the tape tracks. Time codes that are perfectly in sync with the audio signal (but do not in any way alter its quality or content) can be recorded and played back, and time codes can be accurately read even during high-speed search.

In addition, because of the 4-head technique employed in this recorder, the audio signal and time code can be recorded separately. The four-head system also permits tape monitoring, much as you would do with a three-head analog tape recorder. In the Model D-20, the time code only in the sub-data area can be re-recorded by using an external time-code generator. This function can be used to re-record a continuous time code which has become discontinuous because of editing, etc.

Punch-in/punch-out function is also possible thanks to the four-head construction, 2 channels of signal processing circuits, and the digital cross-fade circuits. The cross-fade time between the input signal and the playback signal is set to 15 milliseconds, making smooth punch-in/punch-out possible.

Tape speed is variable over a ± 10 percent range and in 0.1 percent steps. Operation of the tape deck can be synchronized with video (V-Sync) and other digital audio equipment. A specific point on a tape can be located by the absolute time recorded onto the tape as the address for locating. Two locating memories are available for this function. Two-times over-sampling digital filtering is used for A/D and D/A conversion. There are several more useful features available on this professional DAT mastering recorder, but they will be mentioned as we review the controls that bring them into play.

CONTROL LAYOUT

The power on/off switch of this recorder is located at the lower left of the panel. The cassette slot, eject button and the more familiar tape transport buttons (record, stop, play, rewind, and fast forward) are at the upper left in an area set off from the rest of the panel by its dark, contrasting color. A "Shift" key to the right of this area makes it possible to control fifteen functions with just eight pushbuttons nearby. When the shift key is not depressed, these eight buttons handle "Hold & Edit", "Repeat Playback", "Down" and "Up" keys for retarding or advancing time and tape speed numbers on the display, Zero, 1 and 2 locate point keys and a play locate key. Alternate functions of these keys when the shift key is depressed include audible cueing during fast-wind (at five times normal speed), repeat functions, blank search (searching for the last point recorded on a partially recorded tape), and variations on the locate functions described earlier.

A "Display" button alters part of the display to its right to show either data being edited, program number, time

Figure 3 (B). THD versus input recording level in dBm, using the analog inputs.

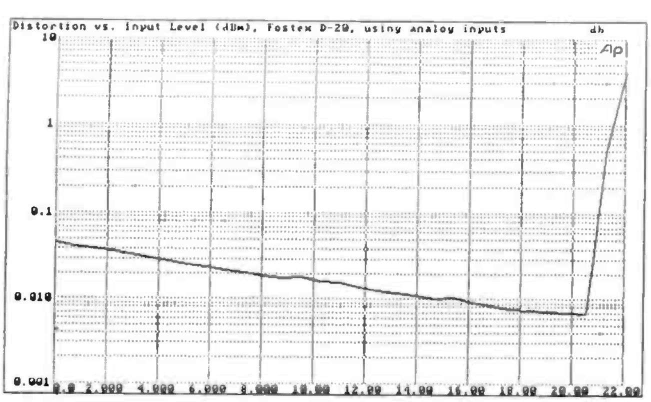
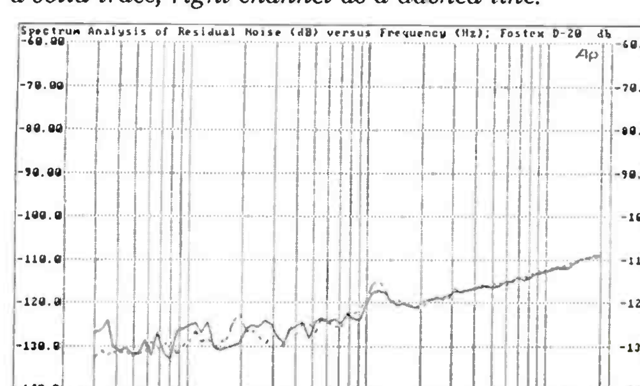


Figure 4. A spectrum analysis of residual noise, when playing a no-signal track of a CD-1 test tape through the D-20 recorder. Left channel is shown as a solid trace; right channel as a dashed line.



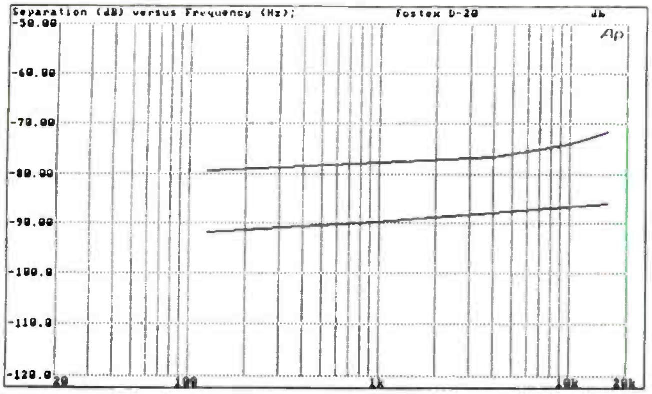


Figure 5. Separation versus frequency. The lower trace is a plot of right-to-left separation; the upper trace shows left-to-right separation.

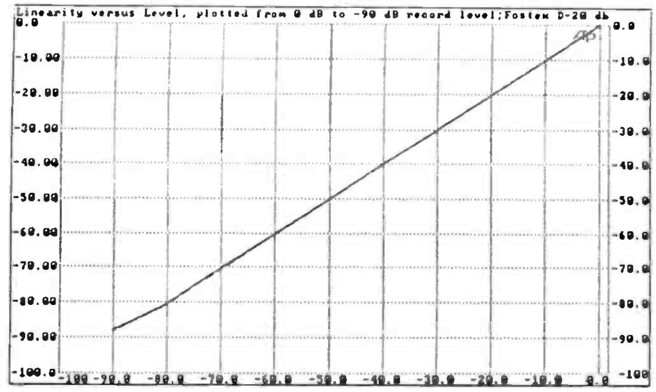


Figure 6. Linearity of the Fostex D-20 D/A converter system, using undithered signals, from 0 dB (maximum) record level to -90 dB.

on the tape or present tape speed. The display area at the upper right of the panel also has a pair of bar-graph level meters calibrated from 0 dB (maximum record level) down to below -50 dB. Below the display area are a master input level and balance control, audio and time-code ready buttons, audio and time-code input monitoring buttons, and audio and time code "Repro" (monitoring) buttons.

Additional indicator lights and switches along the lower edge of the panel include emphasis on/off, copy guard on/off, digital/analog input selector, 44.1 or 48 kHz sampling selector, external/internal clock selector, edit/normal record mode, input/repro monitor selector and uncalibrated/calibrated input level selector switches. A headphone jack with its independent level control completes the front panel layout.

The rear panel is equipped with pairs of XLR (balanced) analog audio input and output connectors, XLR time code and digital input and output connectors, and a pair of unbalanced analog output phone jacks. External sync input and output terminals are also found here. Three DIP switches on the rear panel are used to set up frame frequency and sync mode. Full instructions for setting up these switches are given in the comprehensive owner's manual supplied with the recorder. Finally, the rear panel also has a Data Communication receptacle for use with external equipment that is compatible with RS-422A specifications. A detachable heavy duty line cord is plugged in at the rear panel, which also incorporates a grounding terminal.

LABORATORY MEASUREMENTS

The *Vital Statistics* chart at the end of this report summarizes all of the measurements we made of this impressive DAT mastering recorder. Measuring the performance of any digital recorder that has both digital and analog inputs requires two types of measurements of some of the performance characteristics. For example, the playback frequency response of a *digitally* recorded sweep of frequencies may be different from the response measured when signals are applied via the analog stereo inputs. In the first case we are really measuring the D/A conversion stages and the analog output stages. In the second case we are measuring both the A/D conversion system used to make the digital recording on tape and the D/A elements, plus other playback circuitry.

Accordingly, *Figure 1(A)* shows the response obtained when we played back a special digitally recorded DAT tape that we have in the lab. Response varied by no more than a couple of tenths of a dB from below 20 Hz to around 15 kHz, with a rise of between 0.3 and 0.4 dB (depending upon which channel was measured) at 20 kHz. By contrast, when we used the analog inputs to go through the complete record/play cycle, we saw a slight dip of about -0.3 dB at around 10 kHz, followed by a rise to about +0.25 dB at 17 kHz and a steep roll-off beyond that frequency. Response was down -3 dB at 20 kHz, as shown in *Figure 1(B)*.

Figure 2 is a plot of total harmonic distortion plus noise versus frequency, during playback of our digitally recorded test tape. At 1 kHz, THD plus noise was a mere 0.003 to 0.004 percent, rising to around 0.04 percent at 20 kHz. *Figure 3(A)* shows how THD plus noise varied as a function of recorded level. The graph is plotted in dB below maximum (0 dB) record level. Over most of the range shown, THD plus noise was around -94 dB which is equivalent to about 0.002 percent. At maximum recording level, it increased to between -88 and -90 dB, corresponding to around 0.003 percent. The slight increase is probably attributable to the analog output stages rather than to any problem with the D/A conversion system.

By contrast, *Figure 3(B)* shows the "brick wall" limitations of any digital recording system. It is a plot of distortion versus input level, when input signals were applied to the analog inputs over the range from 0 dBm to 22 dBm. Slightly above 20 dBm, the maximum possible digital code has been reached and any attempt to increase record levels beyond that point results, of course, in a steep rise in distortion.

Signal-to-noise ratio of the D-20, when playing back a digital-zero recorded tape, was an impressively high 104.2 dB on the left channel and 103.9 dB on the right channel, referred to maximum (0 dB) record level as indicated on the unit's own level meters. A spectrum analysis of the residual noise content during this playback is shown in *Figure 4*. *Figure 5* is a plot of stereo separation between channels over a wide range of frequencies. While separation was more than adequate, we were somewhat surprised to find that separation from left-to-right was considerably lower than separation from right to left; just short of 80 dB at 1 kHz for the former case and about 90 dB in the opposite direction.

One of the most important characteristics of any DAT recorder is its low-level linearity. *Figure 6* shows this characteristic over a level range from 0 dB (maximum digital recorded level) down to –90 dB. Linearity was virtually perfect down to –80 dB, and deviated from perfect linearity by about +2 dB at –90 dB.

CONCLUSIONS

Fostex has enhanced the DAT format standards, as originally outlined by a series of International DAT Conferences, in many significant ways that will appeal to professional users of this type of equipment. It is clear that Fostex's experience as a supplier of recording studio equipment has added those features and capabilities that are lacking in most so-called "professional" DAT recorders that are, frankly, not much more than consumer DAT machines that have been fitted with balanced XLR input and output connectors. The Fostex D-20 goes far beyond that, what with its time-code facilities, its four-head construction that permits sophisticated editing as well as real-time monitoring of a recording as it is being made and much more.

Regrettably, we had no way to check out the time-code facilities, since we do not possess an external time-code generator in our lab. A hands-on evaluation of the D-20 by Clay Hutchinson of Cove City Sound Studios will detail the effectiveness of the time-code related features of the D-20. As for our own reaction to this machine, we can only say that any recording studio that has previously avoided DAT as a mastering recorder should be delighted with what Fostex has done to bring DAT up to true professional-use standards. db

VITAL STATISTICS SPECIFICATION

MFR'S CLAIM	db	MEASURED
Recording time (Max.)	120 minutes	Confirmed
Fast For./Rewind (2 Hr. Tape)	80 Seconds	78 Seconds
Sampling Frequency	48 or 44.1 kHz	Confirmed
Dynamic Range	90 dB	94.6 dB
Playback Freq. Range	20 Hz to 20 kHz	±0.3 dB
Harmonic Distortion	<0.05%	0.04% @ 20 kHz
Crosstalk	80 dB at 1 kHz	78/90 dB
Rated Output Level	+4 dBm, balanced	Confirmed
Rated Input Level	+4 dBm, balanced	Confirmed
Power Requirements	120 V,50/60 Hz, 60 W	Confirmed
Dimensions (WxHxD, in.)	19 x 6 x 18-½	Confirmed
Net Weight	33 lbs. (15 kg.)	Confirmed
Shipping Weight	44 lbs. (20 kg.)	Confirmed
Suggested Price:	$8,000.00	

CLAY HUTCHINSON and DAN HETZEL

The Fostex D-20 DAT Recorder: Time-Code Features

• We recently had the opportunity to test synchronization of the Fostex D-20 DAT recorder, using the Fostex 4030 synchronizer and 4035 controller. Our 50-track lockup consisted of two Studer 24-track analog tape recorders, an A-820 and an A-80, and the Fostex D-20. Using the A-820 transport as the master controller, we slaved the A-80 to the A-820 using a pair of Lynx time-code modules, and then slaved the D-20 to the A-820 with the 4030/4035 combination, using the 4035 remote to control the D-20 transport.

While the present Fostex 4030/4035 manual contains no information on synchronization of their new DAT recorder, the D-20's manual filled the void with a section devoted to this. The calibration of the D-20 and the entering of appropriate parameters into the 4035 controller was a simple matter. Once completed, the entire setup locked perfectly and consistently in about eight seconds. Once sync-lock was achieved, it was maintained until otherwise directed by the master controller.

THE UNIT'S VERSATILITY

The initial pass of recording went well, as expected. However, a change in the middle of the mix became necessary, which usually would require razor blade editing on an analog machine. Since this is not possible with DAT cassettes, mixes on conventional DAT recorders must be done in one pass. This is where the beauty of this machine and its sync capabilities become wonderfully apparent. Our mix was adjusted, the tape was started, and at the point at which the change in the mix was to be recorded, the D-20 containing the master mix was punched-in, still remaining in sync with the analog multi-tracks. It went smoothly.

Upon playback of our DAT master, we confirmed that a perfectly *inaudible* punch had been made! In fact, all punch-ins *and* outs were flawless, even those made deliberately and experimentally in less-than-ideal places. The whole setup performed better than we thought possible.

The 4035 remote was a welcome inclusion in our setup and though not used as a master controller in the application described above, it performed very well when tested, controlling a single Studer A-80 multi-track, slaved to the D-20 master. The unit not only has the ability to control a master and up to three slaves (only additional 4030 modules would be needed), it also contains ten memory locations, defeatable auto-play, auto-return, auto-record (including a handy rehearsal mode),as well as many other useful features, most of which are accessed with the touch of a single button. A large LED display contains SMPTE code read from either the master tape machine, a slave, or the offset or time between the positions of the two machines.

While the 4035 would be well-suited as the control center of many different arrangements of synchronized audio and video tape machines, we used it mainly for D-20 playback, utilizing its memory functions and auto-play features mentioned above. However, we found the tape transport controls to be small and stiff compared to the larger, more comfortable buttons on the D-20, so those latter were used for punches. Also the subframe section of the SMPTE display is somewhat misleading. A frame is comprised of 100 subframes, usually numbered 00-99, but the subframe readout here only contains one place, positioned in what is typically the tens place. If this is what it reads, then the display incorrectly suggests seven (7) subframes when it is, in fact, reading seventy (70). In our usage, though, it did not affect the operation of the system in any way.

To our surprise, SMPTE time-code connections to the 4030 needed to be adapted from the balanced XLR-type jacks on the D-20's time-code output and on our multi-track, to unbalanced RCA-type jacks at the back of the 4030 unit. This is a shortcoming, because the absence of properly electronically balanced connections could result in a wide range of audio and SMPTE lockup problems. Also, neither synchronizer unit generates time code, so there was a question of whether the level stripped onto the D-20 was acceptable to the 4030 reader, and for that matter, to the D-20, since LEDs show only the presence of time code on tape, not level. Printing and reading a level of about –7 dBm worked fine with both the D-20 and the 4030.

SUMMARY

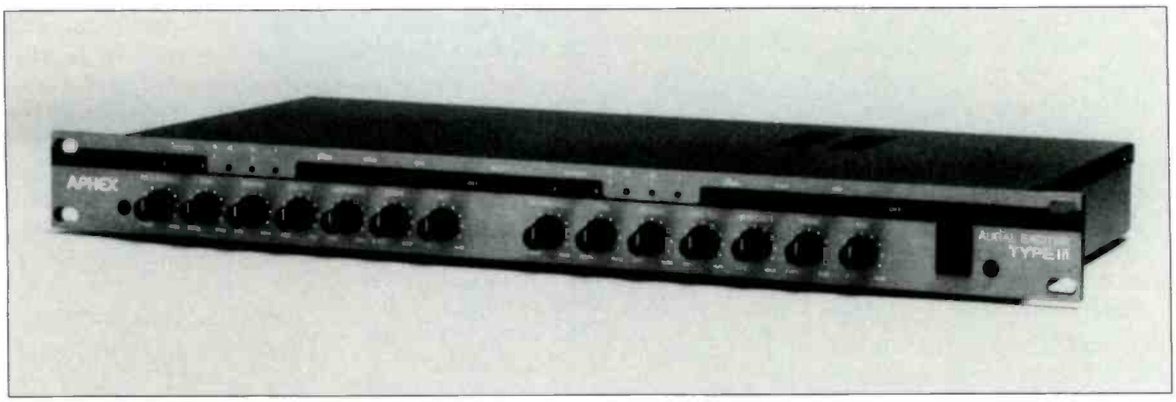
Despite this nitpicking, we found the Fostex D-20 and the 4030/4035 synchronization system to perform superbly in a professional recording studio environment. The superior sound quality and convenience of an all-digital master recording, due to the editing capabilities of the D-20, will likely find usefulness in many areas of audio and video production. The D-20's unique features have been implemented in a package that clearly is ahead of its time. db

Clay Hutchinson and Dan Hetzel operate Cove City Sound Studios in Glen Cove, NY.

JOHN BARILLA

Hands On

Aphex Aural Exciter Type III Model 250



• It was around 1977 when I first heard of a device called the Aphex Aural Exciter. Back then, Aphex had stirred up waves of controversy with this radical new signal processor. We had all heard the hype generated by the trade magazines, but precious few engineers had gotten their hands on one. As far as the rank and file were concerned, use of the Aural Exciter was limited to the audio elite: those who were working large record company-backed projects.

At that time one could not purchase one of these Aphex units—not for any amount of money—for Aphex was not selling them. Instead, they had a policy of leasing the rights to use an Aural Exciter at high fees for discrete periods of time. While this rather aristocratic attitude seemed fairly abhorrent to most engineers who were deprived of access, it seems in retrospect one of the cleverest, most well orchestrated long-term marketing schemes ever conceived. After all, it had accomplished it's goal: it had provoked interest in the product without having to pay for advertising.

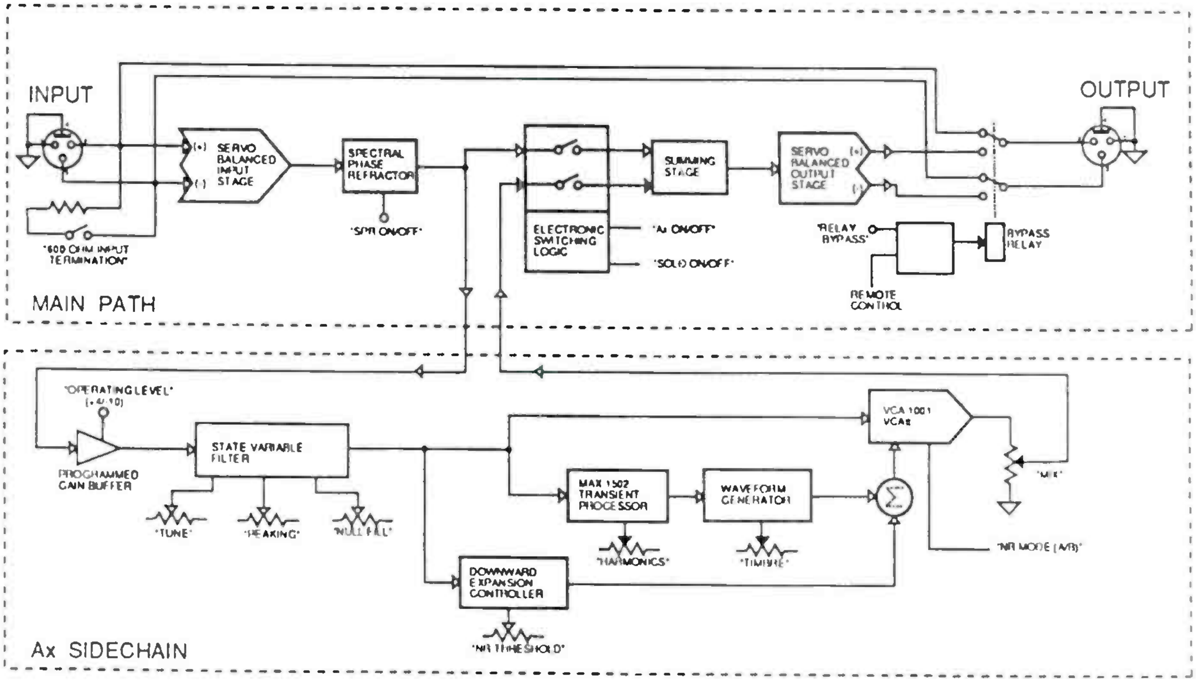
I remember hoping that someday we would be able to get our hands on one. But until then we peons would have to content ourselves by listening to hit records reportedly processed with the Aural Exciter. Had Aphex invented the dream machine? It might have seemed that way at first since all the records bearing the Aphex stamp would go gold. Or was it that Aphex had the incredibly good market savvy to associate the Aural Exciter with only top selling recording artists?

Back then, when engineers and producers got together for coffee or beers, the conversation invariably turned to the Aphex issue. "Did you hear the Jackson Browne record? Did you sense anything different about it? How would you describe what you heard?" The ensuing discussion often got hot and heavy as the armchair audiophiles attempted to dissect the essence of the Aphex process from its results on a disc.

Predictably, the arguments became extremely polarized. Some said it sounded like nothing more than distortion added to an otherwise clean recording. Their argument was a typical *reductio ad absurdum*: "why bother watching your record levels if distortion is such a desirable attribute". To them, the very notion of adding a controlled amount of harmonics to a scrupulously recorded master tape was ridiculous beyond belief. For these engineers, the Aural Exciter was the ultimate scam—the triumph of public relations techniques over common sense.

On the other hand, I was a lot more conciliatory about the device. My philosophy of audio had always been nonconformist. So I was open. I had heard the results and was convinced that Aphex-processed recordings did have more impact on the listener. Just how this was accomplished was not so significant to me as it was to the "techies". What was significant was that it worked. Whereas multi-track recording in the mid 1970's—with its strict isolation, dead rooms and close mic techniques—had become sterile and lifeless, Aphex processed recordings seemed to have the realism and impact restored to them—even if they had been overdubbed into oblivion.

By 1979 Aphex was no longer marketed as an exclusive-use item. They were now selling Aural Exciters to



NAMES ENCLOSED IN QUOTES ARE OPERATOR CONTROLS
ONE CHANNEL OF TWO IDENTICAL CHANNELS IS SHOWN
▷ = SIGNAL PATH
▽ = AUDIO GROUND

Figure 1. Block diagram of the Aphex Type III Aural Exciter.

anyone who could afford one and an audio dealer in New York offered to let us hold onto a unit for thirty days to evaluate it. During that period of time, even the skeptics in our studio came to respect the Aphex concept as a legitimate one. As for me, I processed a number of my recordings with the Aural Exciter. Now ten years later, when I play even cassette copies of these recordings, they sound extremely fresh compared with unprocessed tapes from the same period.

Aphex has come a long way in the past dozen or so years. The Model 250 is the third version of Aphex's proprietary technology. While the principles of the Aural Exciter remain the same, the technology by which it is generated and controlled have grown greatly in sophistication. The mysterious looking black box with only a few primitive controls has now been replaced by a precision set of parameters that allow the user to fine-tune the exciter for a variety of applications.

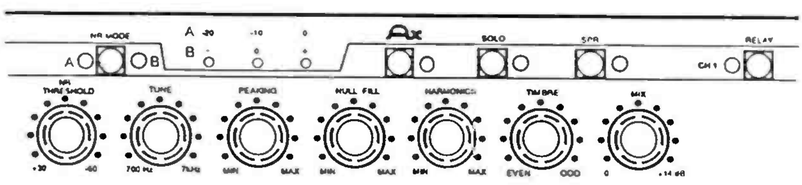
THE HEART OF THE APHEX

While there are a whole bunch of important parameters that can be controlled from the front panel, two of the controls—*harmonics* and *timbre*—are what essentially defines the Aphex process. They are really the heart of the Aphex Aural Exciter. What the Aphex does is generate harmonics related to the input signal and fold a portion of it back into the main signal. This proprietary process does something that equalization never could do—add harmonics where there were previously none. Without an Aphex you can add some gain to a weak harmonic with EQ (and simultaneously add unwanted noise). No amount of EQ can bring out a harmonic that does not exist, but that is precisely what an Aphex can do. Adding harmonics can be very useful in extending the bandwidth of a recorded sound so that it becomes sonically "alive." Likewise, it can also be used in some perverse, but musically interesting ways as well.

Early Aphexes generated harmonics rather primitively by rectification. It was an effective means, but sometimes difficult to find the proper setting. In the new Aphex Type III harmonics are generated "intelligently." They are no longer the products of harmonic distortion but rather the product of a transient processor and waveform generator. Hence the harmonics have less inter-modulation generation, which means a cleaner, more controllable sound.

Since these harmonics are synthesized directly in response to the envelope of the input signal, there is no in-

Figure 2. Front panel controls of the Aphex Type III Aural Exciter.



crease in overall gain as the amount of harmonics is increased. The result is that this new Aphex is a lot more forgiving when it comes to settings. It inclines the user to experiment without having to reset all the other controls. Additionally, the input/output structure of the unit is designed so that unity gain prevails—no matter what the settings.

While the *harmonics* control increases the harmonic content as the knob is turned clockwise, the *timbre* control allows the user to choose between even order harmonics, odd order harmonics or any combination thereof. Adding even harmonics can give your track (or program) a nice warmth, whereas the odd harmonics will give it some more edge.

In a tracking situation, the Aphex can really save you a lot of time in trying to come up with that quintessential "hot" drum sound.

TUNING IN

Even though the harmonic and timbre controls are described as the heart of the Aphex, they don't work in isolation. Provision has been made for tuning-in the range of frequencies over which the harmonics will be generated. Three controls act together to pre-condition the signal before it hits the harmonics generating section: They are labeled *tune*, *peaking* and *null fill* (see *Figure 1*). As parameters of a state-variable filter, these three controls allow the user to determine, in a very flexible way, the area of the incoming signal which will be most appropriate to the action of the harmonics-generating section.

The *tune* control sets the frequency range of a high-pass filter. Only those frequencies above the user-determined cut-off point will have any effect on the harmonics generator.

To further accentuate the effect, specifically about a narrower group of frequencies, there is also a *peaking* control. Here the user can design into the high-frequency shelf (which was already established by the tune control) a resonance peak right about the cut-off point of the high-pass filter. Not only does this put a peak in the sound, but also a null in the frequency range just before it, allowing you to create a nifty little spike at one given area.

If this emphasis seems to be a little excessive, the null area can be literally back-filled using the *null fill* control. With controls of this sophistication, a user can turn the Aphex Aural Exciter into a very "personalized" instrument.

MIXING AND SWITCHING

Whatever sound enhancement you construct with EQ and harmonics generation can be utilized in two different ways. If the sound (an individual instrument or a program) is going directly through the Aphex, you can use the on-board mix control to fold it back into the main signal. On the other hand, if you want to "excite" different tracks at different levels during a mix you will need to use an effects bus to drive the Aural Exciter. Aphex obliges us here by including a solo switch which returns 100 percent of the excited signal to the effects return at the console. Speaking of switches, there are several of them on the face plate of the Aural Exciter. The AX switch takes the exciter in and out of the circuit but leaves other processing activated. The relay switch however, provides a hard wire bypass of every function in the unit.

One of the more interesting features of this new Aphex unit is the *SPR*—Spectral Phase Refractor. Engaging this switch apparently realigns the phase relationship of low, mid, and high frequencies, so that the lows lead the mids and highs. The aim is to compensate for the inevitable bass "smearing" that occurs due to the processes of recording and duplicating. This slight lead seems to make up for the naturally occurring delay in low frequencies which occurs during the recording process. Audibly, when switched in and out of the circuit, the effect of the SPR is subtle—as it should be, but bass instruments seem to "tighten-up." If this preserves the integrity of phase response for several tape generations it is indeed a very useful tool.

NOISE REDUCTION

Aphex has chosen to include a comprehensive noise-reduction system in this Type III exciter. In fact, they have included two modes of noise reduction which work on different principles and are suited for different applications.

Why do you need noise reduction in an exciter? Because when you selectively EQ the high end of a sound and then generate harmonics on top of that, you can't help but generate noise along with it. Even with the harmonics generating section set to a minimum, the effects of the boost in EQ alone can produce hiss. Couple that with noisy sound sources or hiss from tape, and you've got a real problem on your hands.

Aphex deals with this problem quite effectively with their two noise-reduction modes. It is worth noting that while the user can switch between noise reduction *mode A* and *mode B*, it is not possible to switch noise reduction out of the unit's side chain. Evidently, Aphex has sought to assuage the critics who dismiss aural excitation as "too noisy" a process. That criticism might be valid if the "side-chain" (that section of the Aphex which does the exciting) remained wide-open constantly. In that case, there would be a subtle, but perhaps annoying drone of high-frequency trash emanating from the side-chain irrespective of whether a signal was loud or soft.

Aphex's solution is simple, but elegant. If you simply turn down the exciter's side-chain when the source signal drops below a certain threshold, there will be no hiss to hear. And while the signal is hot, any hiss will be insignificant relatively to the magnitude of the sound. That, in a nut-shell, explains Aphex's *Mode A* noise reduction. It is essentially a linear expander inserted on the output of the side-chain prior to the mix control. Utilizing a gentle 1.25:1 ratio, the unit tracks smoothly over a 100 dB range.

While noise reduction *Mode A* is nothing but good audio common sense, *Mode B* is something that seems to be innovative and unique to the Aphex system. *Mode B* acts similarly to the *Mode A*'s downward expansion to a point about 8 dB below threshold. At this point the side chain is just about eliminated from the mix.

But here's the new twist. Below this point, the side chain begins to be re-integrated into the mix, but this time it is 180 degrees out-of-phase with the main signal. The result is that some audio broad band noise can be effectively canceled—actually erased—from the main signal. While I had no way of quantifying this feature, it is clear that some noise can be eliminated from the source in this manner. Aphex reports that up to 5 dB of cancellation is to be expected on appropriate sources.

What are appropriate sources? Well, *Mode B* is tailored for use with single instruments, particularly those with strong initial transients like drums. Individual tracks off tape should yield good results here. This feature is not helpful on complex program material. For fully-mixed programs, *Mode A* is your best choice.

APPLICATIONS

So what can you do with this gadget? Lots of things! You are really only limited by your willingness to experiment. Perhaps one of the reason why Aphex Aural Exciters are not as popular as digital reverbs is simply because you just can't push a button and automatically have the optimum sound. You have to play with the unit until you've maximized its potential for that particular sound or program. But if you've never used an Aphex before, you would probably be pleasantly surprised to find out some of the many applications it's suited for.

In a tracking situation, the Aphex can really save you a lot of time in trying to come up with that quintessential "hot" drum sound. If you process a drum sample through an Aphex you'll be amazed how quickly you can turn a static snare or kick drum sample into a lethal weapon! It's easy to put some "crack" into a snare or add a nice clean click to the kick drum without getting into radical EQ. In the end, the sound will be much cleaner with the Aphex, because a little processing goes a long way. Many synth sounds and even acoustical guitars can also be made more life-like by the addition of some upper harmonics.

Much use of the Aural Exciter can be heard on recordings of many well known pop vocalists. If you notice a certain brilliance and airiness that makes the vocals very intelligible despite being nestled into the music, you have probably heard the Aphex in action. In my own experiments with this particular unit, I was actually amazed at the cleanness and discrete tunability of the harmonics. I processed a male vocal with it and came up with a sound I was well pleased with. But when I really listened closely to what was happening, I thought I could hear a female vocalist tracking the male vocalist, faintly but perfectly in the background. I was certain that I had not used a harmonizer on that channel. Upon switching the exciter out of the line, the female "ghost" disappeared. Of course, if I had not liked the effect I could have backed the Aphex off to a more subtle level and still benefited from it. But I did like the effect so I used it! Generating a second harmonic in this manner would have been impossible with EQ or with any of the other exciter-type units on the market which do not utilize Aphex's patented technology.

As I mentioned earlier, a little Aphex goes a long way. So with vocals in particular, it might be better to use the Aural Exciter as a mix function so that you can dial in just the right amount without guessing. Drums and other percussive instruments however, are more safely excited on the way to tape. So if you only have one unit, you can use it several different points in the recording chain.

With the original Aphexes—(the prehistoric black boxes I spoke of earlier)—that was a real "no-no." Because they didn't have the sophisticated tuning section now available, it was very possible to get too similar of a harmonic build-up on several tracks, thereby fatiguing your ears. But with the state-variable filter section of the new Type III Aural Exciters, it is much easier to make liberal use of the exciter on tracks and also on the entire mix.

The Aphex can be used on the overall mix in several ways. One way is to treat it as an effect, sending and returning it through a bus. You can do this by placing the Aphex in solo, which returns only the side-chain (the processed signal) to the console. This has advantages in that each instrument can be selectively excited to different degrees, and of course through the console various collateral treatments of the Aphex can be done, such as cross-panning, flanging or some other effect.

The one drawback to doing this is that the SPR (Spectral Phase Refractor) switch does not seem to be too helpful when the Aphex is placed on a bus. In fact, it seems best to shut it off in that sort of application, because it may "confuse" the phase relationship of the various tracks, depending on how much of each track is being fed to the Aphex.

The SPR can, however, be used to great benefit if the entire stereo mix is fed through the unit on it's way to the two-track mixdown machine. Here, you would not be in solo, but rather utilize the mix controls on the face of the Aphex. Here, the SPR can be useful in making the bass frequencies more coherent. Subjectively, it seems to draw bass information inward—towards—that imaginary center channel. The SPR may not always be appropriate though. If you're going for a big, "woofy" synth bass sound (as I often do) the SPR may be counter productive. But it's nice to have the option!

One of my favorite applications though, is simply in making cassette copies. Don't abuse your master mixdown by running it every time you want to make a copy. Instead, just make an Aphex processed secondary "running" master—optimized for tape duping—and put your original back on the shelf. If you do this properly, you'll find it very easy to make clean, powerful tape copies that sound virtually identical to your original master.

CONCLUSIONS

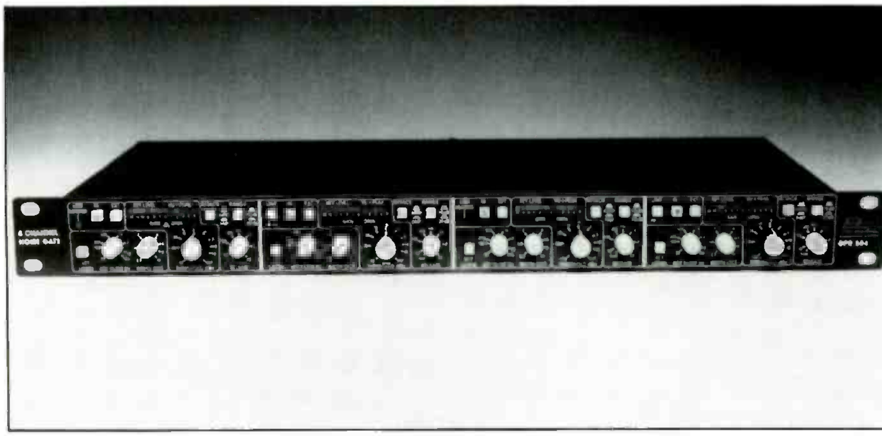
Every studio may not yet have an Aphex Aural Exciter Type III, but many should consider buying one. Aural excitation is no longer a controversial item as it was in the 1970's. It is a proven item, as many hit records, tapes, broadcast and sound reinforcement applications have demonstrated. Through a decade and a half of use, and three technical revisions, the Aural Exiter has shown itself to be a unique and valuable signal processor. It has earned a place in every studio rack—right next to the compressor, gate and reverb.

List Price: $995.00

db

New Products

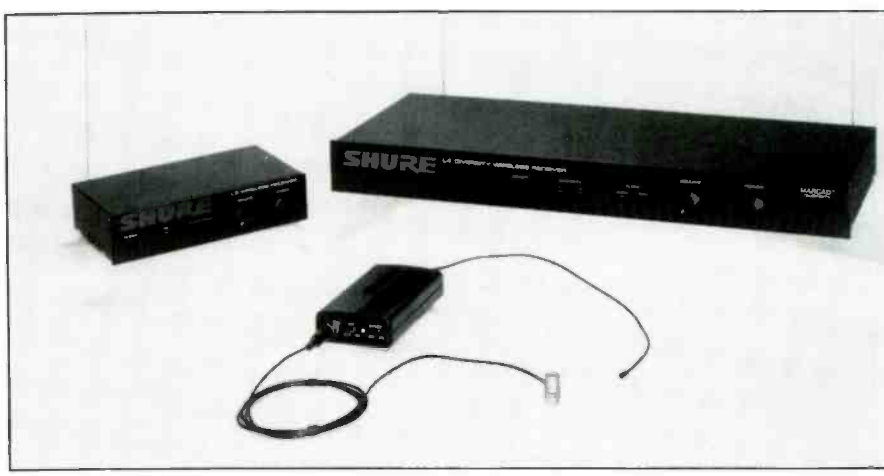
NOISE GATE



• The DPR-504 Four Channel Noise Gate is a new unit ideally suited for cleaning up multi-miked drums or percussion setups, eliminating spillover from backline instruments and monitors on stage, as well as from adjacent percussion instruments. It is also useful in live sound mixing to highlight instruments such as lead and bass guitar, while still keeping amplifier noise down during quiet passages and breaks.

Mfr.—BSS
Price: $1,999.00
Circle 31 on Reader Service Card

WIRELESS MIC SYSTEM



• L Series products are being offered in four complete systems developed to suit varying user needs. For guitars and instruments, System LS13 (with an L3 Non-Diversity Receiver) and System LS14 (with an L4 MARCAD Diversity Receiver) both include the L1 Body-Pack Transmitter and the WA300 Instrument Adapter Cable. The LS13/839 and LS14/839 Wireless Lavalier Systems consist of the L1 Body Pack Transmitter, the L3 or L4 Receiver, and the 839W Omni-directional Lavalier Microphone. A noiseless audio mute switch allows cut-off of sound from the instrument or microphone without shutting off the transmitter, and the L1's TA4M "universal" input connector permits use of the transmitter with other compatible microphones and electronic musical instruments. The 839W Wireless Lavalier Condenser Microphone's frequency response is specially tailored for lavalier applications with a controlled low-frequency rolloff to reduce pickup of unwanted noise. The side-exit cable and special tie-bar mounting accessory make the 839W visually unobtrusive. The 839W (and other Shure wireless microphones) use a TA4F "Tiny Q.E." connector for direct hook-up to the L1 Transmitter. The L Series L3 Non-Diversity Receiver offers a durable, all-metal case that is professionally styled and rack mountable. A detachable, quarter-wave whip antenna with a UHF-type connector can be remote located for rack installations and for difficult pickup situations. A red Audio Peak warning light indicates audio overload. These and many other features, including double-tuned RF stages, dual ceramic IC filters, and a three-pole Chebyshev audio low-pass filter, assure clear, professional-quality sound. The L4 MARCAD Diversity Receiver is equipped with all the features of the L3 Non-Diversity Receiver plus a balanced mic. level output. It can be rack-mounted, and it comes with two detachable, quarter-wave whip antennas with UHF-type connectors. Designed for professional applications, the L4 MARCAD Diversity Receiver can be used in any demanding audio situation.

Mfr.—Shure Brothers, Inc.
Price: $360.00 (LS13)
$495.00 (LS14)
$445.00 (LS13/839)
$580.00 (LS14/839)
Circle 32 on Reader Service Card

DIGITAL DISK WORKSTATION

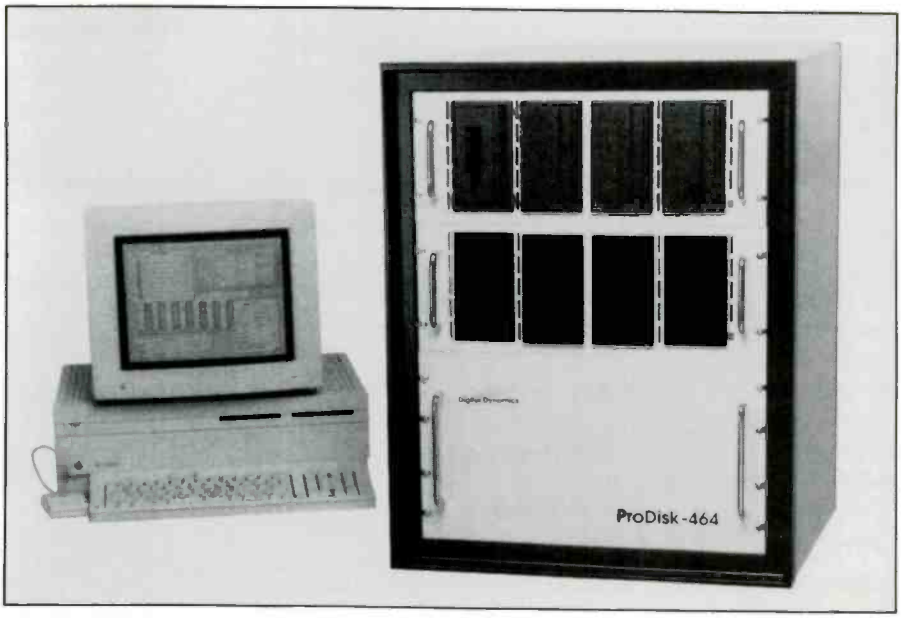
• A new economical 4 to 64 track digital-disk random access audio recording and editing workstation is now available. Engineered for music recording, video/film post production, television and radio broadcast applications, the ProDisk-464 features high speed Winchester disk storage, plug compatibility with multi-track tape machines, 96 dB dynamic range and a 90 dB signal-to-noise ratio. The system delivers less than 0.02 percent, 20 Hz to 20 kHz total harmonic distortion, is SMPTE time code and MIDI compatible and uses familiar editing techniques presented on Apple Macintosh screens.

Mfr.—Digital Dynamics, Inc.

Price: $25,995.00 (starting price)

Delivered 21 days A.R.O.

Circle 33 on Reader Service Card



LOUDSPEAKER

• Model C12A is new coaxial loudspeaker consisting of a 12-inch diameter low frequency woofer and a 4-inch diameter piezo high

frequency driver. Amplifier output wiring connections are by screw terminals, and a mounting bracket and hardware are provided for convenient installation of line matching transformer. The loudspeaker has a frequency range from 50 Hz to 19,000 Hz and a power handling rate of 65 watts. It is designed for optimum performance when mounted in enclosures with a volume of 3 cubic feet.

Mfr.—Atlas/Soundolier

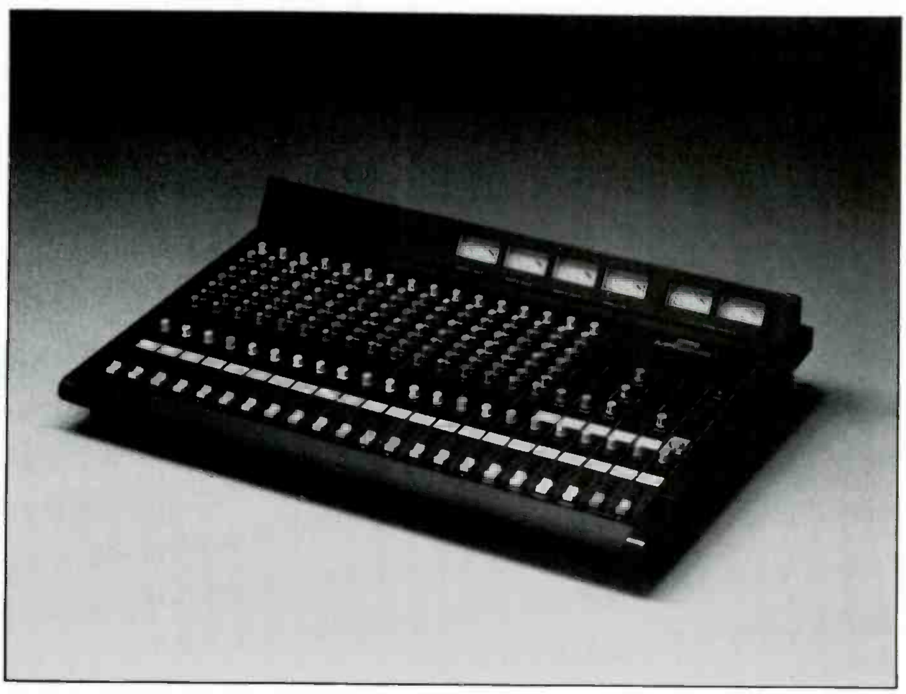
Price: $268.72

Circle 34 on Reader Service Card

MIXING CONSOLES

• Available in three different input configurations, the MR Series is ideal for recording studio use, sound production applications, and small sound reinforcement needs. Each MR Series Console has four mixing buses and a stereo master bus. They are available as the: MR842 with 8 inputs, the MR1242 with 12 inputs, and the MR1642 with 16 inputs. Each input features a choice of electronically balanced, low-impedance, XLR-type inputs, or balanced TRS phone jack inputs, with built-in, phantom power for condenser microphones. Each input feeds a switchable 20 dB pad to a continuously variable input trim control to a precision long-throw fader. Each channel includes a 3-band equalizer with ±15dB boost/cut and sweepable peaking mid-frequency control. There is also a pre-EQ insert jack and each input has a peak overload indicator for precise level optimization.

Mfr.—Yamaha Corp.



Price: $1,295.00 (MR842)

$1,595.00 (MR1242)

$1,895.00 (MR1642)

Circle 35 on Reader Service Card

DIGITAL CART RECORDER/REPRODUCER

• The model 9500 is the first available from the Series 9 Digital Broadcast Cart Machines, The Model 9500 offers 16-bit digital performance, and provides three cue tones, looping, separate gain controls, actively balanced inputs and outputs, a parallel remote, and SCSI port. VU and peak LED metering is standard. The model 9500 will perform as a stand-alone recorder/reproducer. An optional personal computer or terminal connected to the serial port will access advanced functions.

Mfr.—Ferrograph—Gotham Audio

Price: available upon request

Circle 36 on Reader Service Card

DUAL BIT AUDIO D/A CONVERTERS

• The DAC D20400: Dual 20 Bit Audio D/A Converter-8x Oversampling, and the DAC D18400: Dual 18 Bit Audio D/A Converter-8x Oversampling are now available. The DAC D20400 converts with 20 bit resolution at sampling rates up to 400 kHz. It includes two complete D/A Converters, a stable bipolar reference, a universal serial CMOS/TTL compatible digital interface and two distortion-suppressing output deglitcher amplifiers. The DAC D18400 is a Dual 18 Bit Audio D/A Converter that provides 18 bit performance. Like the DAC D20400, this product also converts at 8X oversampling rates and offers the user a complete solution. Critical performance parameters such as differential non-linearity, S/N ratio, monotonicity, and harmonic distortion are consistent with the product's 18 bit resolution.

Mfr.—UltraAnalog, Inc.

Price/Delivery: (two complete channels)

$325.00 (DAC D20400) (in single quantities)

$209.00 (DAC D20400) (@ 250 each)

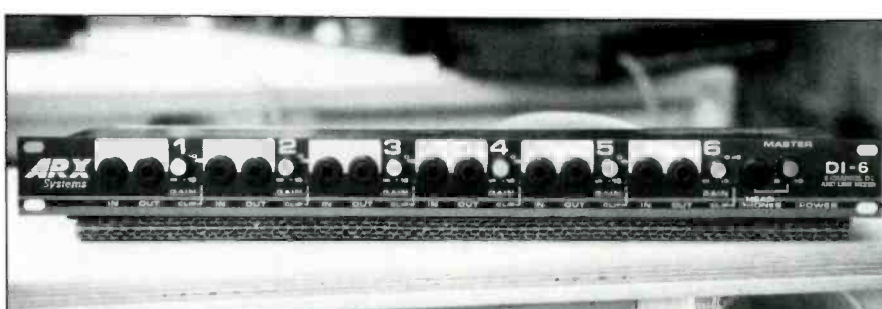
$265.00 (DAC D18400) (in single quantities)

$129.00 (DAC D18400) (@ 250 each)

Deliverv is 2 to 4 weeks A.R.O.

Circle 37 on Reader Service Card

DIRECT BOX/LINE MIXER



• The DI-6 is a six-channel active DI box, which also doubles as a line mixer. This unit allows six independent audio sources to be interfaced with balanced Lo Z professional sound systems; either to individual channels or summed down to one master output. Each channel has 2 x¼-inch (6.5mm) phone jack IN/OUT sockets, level control with up to +15 dB gain, clip LED, and blank numbered panel for easy confirmation of DI assigns. The Master section has a master volume control, and a headphone socket for personal monitoring. On the rear of the unit, each channel has an XLR balanced output, earth (ground) lift switch and indicator LED. The master section has both XLR and Phone jack outputs.

Mfr.—ARX Systems

Price: available upon request

Circle 38 on Reader Service Card

Signal Processing II

In this issue we complete our roundup of what's available in signal processing. This issue covers Compressors, Limiters, Noise Gates and Expanders (and combination units of the preceding), and Noise Reduction Systems.

As usual, however, we must remind you that the information contained is from the respective manufacturers. Further, if a manufacturer you seek is not listed, the chances are strong that, as many times as we tried to get information, we could not get it from them.

COMPRESSORS

APHEX SYSTEMS LTD.
The Compellor (Model 300) is a stereo compressor providing smooth,inaudible compression. It features "intelligent" control circuits which simplify set-up and require no readjustment for varying program material. Fits a single rack unit and weighs 11 lbs.

Price: $1195.00

The Compellor (Model 301) has all the features of the 300, but is mono.

Price: $795.00

LIMITERS

ALTEC LANSING CORPORATION
14712A is a plug-in two-channel limiter for the 9400 series power amplifier providing loudspeaker protection by limiting output power. Has selectable threshold and attack/release controls. Measures 2.75 x 3.60 x 1.25 in. and weighs 6 oz.

Price: $164.00

APHEX SYSTEMS LTD.
The Dominator (Model 700) is a multi-band peak limiter which provides an absolute peak ceiling with zero overshoot. The multi-band aspect is free from "hole-punching". Increases "apparent loudness". "Intelligent" circuits control thresholds automatically. Occupies a single rack space and weighs 8 lbs.

Price: $1195.00

UREI-JBL PROFESSIONAL
1178 Dual peak limiter features two independent peak limiters that can track together in stereo mode. Attack time is adjustable from 20-80 μs; release time from 50 ms to 1.1 sec. Also features four compression ratios and switchable meter ballistics (VU, peak) in a double rack space unit.

Price: $1150.00

NOISE GATES AND EXPANDERS

ALESIS CORPORATION
Micro Gate is a keyable stereo noise gate with threshold, delay and rate controls. Features smooth, quiet operation at 20 kHz bandwidth. Has in/out switch and occupies ⅓ rack space and may be interlocked with other units.

Price: $149.00

APHEX SYSTEMS LTD.
Model 612 Expander/Gate allows true downward expansion with variable ratio, allowing variety of dynamics control. Promises "no clicks regardless of attack speed". Can also be used as a 'ducker'. Occupies a single rack space and weighs 7 lbs.

Price: $795.00

ASHLY AUDIO INC.
Model SG-33 is a 2-channel noise gate utilizing a proprietary VCA exhibiting low distortion, and low control-voltage feedthrough (turn-on click). Dual-release time circuit offers "wait-before-fading" option. Occupies a single rack space and weighs 8 lbs.

Price: Available upon request

Model SG-35 is a 4-channel noise gate exhibiting all the features of the SG-33. It occupies a double rack space and weighs 10 lbs.

Price: Available upon request

ARX SYSTEMS
SixGate offers 6 channels of independent gating in a single rack space. There are front panel controls for threshold, attenuation, release, key, input/detector loop and hardwire bypass. Weighs 5.5 lbs.

Price: $649.00

FURMAN SOUND INC.
Model QN-44 Quad Noise Gate features four independent noise gate channels in a single rack-space unit. Each channel has threshold, attack, release, and depth controls with "channel-on" indicator. Key input jacks are provided for special effects. Weight: 7 lbs.

Price: Available upon request

COMBINATION UNITS

ARX SYSTEMS
Quadracomp is a 4-channel compressor/limiter in a single rack space package. Front panel offers control of threshold, ratio, and output gain. Features 10-segment reduction meter and hardwire bypass. Weight: 5.5 lbs.

Price: $799.00

ALESIS CORPORATION
Micro Limiter is a soft-kneed limiter-compressor whose preset attack rate adjusts automatically according to nature of the input signal, maintaining a consistent output level. Occupies ⅓ rack space. It can be interlocked with other units. Has input, release and output controls.

Price: $149.00

ALTEC LANSING CORPORATION
1712A Compressor/Limiter features electronically balanced in/out, variable threshold from –45 to +20 dB, variable compression Ratio from 1:1 to infinity:1. Maximum compression is 60 dB. THD is 0.05% at 20 dB of compression. Housed in single rack space. Weight: 6.3 lbs.

Price: Available upon request

ASHLY AUDIO INC.
Model CG-85 Gated Compressor/Limiter features independent attack, release and ratio controls, adjustable attack and release thresholds. Has a gated release function to minimize "breathing", and side-chain patch points. Housed in a single rack space, the unit weighs 8 lbs.

Price: Available upon request

Model CL-50 Compressor/Limiter is especially suited for placing a ceiling on peak levels of program material. Detector circuit allows for frequency sensitive limiting. Has "double release time-constant" to minimize "breathing". Occupies a single rack space.

Price: Available upon request

Model CL-52 is a dual-channel version of the CL-50.

Price: Available upon request

DOD ELECTRONICS CORPORATION (AUDIO LOGIC)
R825 is a single channel compressor/limiter featuring a sophisticated de-essing circuit. It is linkable to another unit for stereo operation and allows access to the signal processing side chain.

Price: $259.00

Audio Logic MT66 is a stereo compressor/limiter capable of soft kneed dynamic range compression, or hard or soft limiting from 1:1 to infinity:1, with up to 25 dB of gain reduction, and includes accessible side chains and a noise gate on each channel.

Price: $320.00

Audio Logic MT44 is a four-channel noise gate in a single rack unit space. Release time, threshold, and range parameters are user controllable. Each gate has side chain access and keying ability.

Price: $350.00

FURMAN SOUND INC.
Model LC-X Expander/Compressor/ Limiter has three independently functional sections: expander/gate, compressor/limiter/de-esser, and hard limiter. There are three threshold controls, and two ratio controls. Features side-chain jacks and on/off transient muting in a single rack space. Weight: 7 lbs.

Price: $349.00

Model LC-6 Stereo Limiter/Compressor/Gate is a two-channel unit that may be switched for stereo operation. Has separate compression and gate threshold controls, side-chain jacks and a ground lift switch in a single rack space unit. Weight: 7 lbs.

Price: $419.00

Model LC-3A Limiter/Compressor is an "economical" unit which has input, output, attack, release, and ratio controls as well as side-chain jacks, de-ess button and a ground lift switch in a single rack space. Weight: 7 lbs.

Price: $249.00

LT SOUND
CLX-2 is a feed-forward compressor/limiter incorporating the Allison EGC-101 VCA. Features include simultaneous operation of both compressor and limiter.

Price: $895.00

ACC-2 is similar to the CLX-2 but incorporates an expander as well. An outboard oscillator is included for tremolo and stereo panning.

Price: $1250.00

SK-2 is a stereo limiter/expander with features that include simultaneous limiting and expansion functions, de-essing and stereo or independent operation.

Price: $395.00

ORBAN, A DIVISION OF AKG ACOUSTIC
464A Co-Operator is a gated stereo leveler/compressor/high-frequency limiter/peak clipper. Features include a faster compression function that can be switched in to provide additional transient overshoot protection, a defeatable silence gate to prevent rush-ups and pumping, six switchable high-frequency limiting curves, and two LED bargraphs. Other features include concealed "least used" controls, and a defeatable clipper.

Price: $1,195.00 plus $24.00 with XLR in/out

424A gated compressor/limiter/de-esser features selectable linear or exponential release time characteristics, defeatable gate with adjustable threshold, separate compressor/limiter and de-esser control loops, better than 25 dB de-ess gain reduction, multiple channels can be connected to track together, balanced inputs and outputs.

Price: $1,095.00 plus $24.00 with XLR in/out.

412A compressor/limiter features peak limiting and compressor functions are crosscoupled to eliminate potential pumping and modulation effects, THRESHOLD control with 20 dB range, hard-wired system bypass switch, side-chain externally accessible for special effects, STEREO COUPLING switch for stereo/dual mono operation, RFI suppression on input.

Price: $459.00 plus $12.00 with XLR in/out.

PEAVEY AUDIO MEDIA RESEARCH
AMR CDS2 features compression ratio which is achieved by a single knob adjustment, and the attack/release time variable is governed by a three position switch selector. Also provided is a user variable de-essing control and a five segment LED array to display all valid gain reduction data. The unit may be operated in dual/mono or "link" stereo. The side chain capability on both channels allows for external compression triggering. Insert jack is a 0.25-inch stereo (TRS) connector.

Price: $199.50

AMR NGT2 offers dual channel operation with a side chain feature that allows the channel's gating action to be externally triggered, in addition to a full array of gate parameter controls. Each channel also includes a trigger output (RCA jack) for applying a trigger pulse to external synchronizable instruments such as electronic drum machines, etc.
Price: $199.00

QUAD-EIGHT
CLEG-222 combines the functions of four devices in a 5.25-inch module that can be mounted in-line with a Westar I/O module in an overbridge, or adjacent in a mounting rack assembly. Switches insert the unit at mic-pre, equalizer out, tape out, or patch. Operation is indicated with a 20 segment LED display.
Price: $545.00

CLG-32 dynamics unit replaces the preamplifier of a Westar console for line-in applications, such as MIDI and post-production. Simplified controls allow fast, easy operation, with a single switch-COMPRESSOR/LIMITER used to change the functions.

Price: $400.00

RANE PROFESSIONAL AUDIO

Model DC24 Dynamic controller is a 2-channel compressor, limiter, expander gate system with built-in 24 dB/octave Linkwitz-Riley crossover. Compressor and Limiter work independently with no interaction on threshold settings. The internal crossover allows the unit to separate sound into two frequency ranges and treat them separately. Comes as a single rack space unit weighing 7 lbs.

Price: $549.00

ROCKTRON CORP.

CE-1 Compressor/Expander features 40 dB of gain reduction or downward expansion, gating. It is foot switchable, stereo strapable, and has master-slave capabilities in a ½ rack space.

Price: Available upon request

Model 300A features 40 dB of gain reduction, peak limiting, "Hush" noise reduction, side chain, zero meter output level adjustment and gain reduction metering in a single rack space.

Price: $419.00

Model 311 Compressor/Expander features 40 dB of gain reduction or downward expansion, gating. It is stereo strapable, has master-slave capabilities, input gain control and foot switchable in a single rack space.

Price: Available upon request

Model 321 Compressor features two channels of compression, peak limiting, stereo master, side chain, full gain reduction, metering, adjustable threshold and ratio in a single rack space unit.

Price: $379.00

Model 360 Compressor features two channels of compression, peak limiting, stereo master, "hush" noise reduction, side chain, gain reduction metering, adjustable threshold, and ratio in a single rack space unit.

Price: Available upon request

RSP Technologies' 2200 Multiband Compressor features multi-band compression, leveling, peak limiting, "hush" noise reduction, balanced XLRs, meter "zero adjust" and master-slave.

Price: $899.00

SONIC IMAGE LTD.

Focusrite ISA 130 Dynamics Processor is intended to complement the existing ISA 110 input signal amplifier. It consists of five independent sections: a compressor/limiter, de-esser/exciter, noise gate/expander, gain reduction and signal metering, EQ and filters.

Price: $2900.00

SYMETRIX INC.

Model 501 Peak-RMS Compressor/Limiter includes separate processors for simultaneous compression and infinity:1 peak limiting. It provides absolute overload protection. Balanced and Unbalanced in/outs are featured.

Mode 501-01 is same as 501, but with transformer-coupled output.

Price: $329.00

Model 544 Quad Expander/Gate has four channels, with front panel selection for gate or downward expansion. Includes side chain and external trigger inputs. Has both balanced and unbalanced connections.

Price: $649.00

Model 525 Duak Gated Compressor/Limiter is a two channel or true stereo unit with program controlled attack and release times. While compressor/limiter governs levels, expander/gate reduces extraneous sound. Has side chain accesibility.

Price: $495.00

Model SX206 Multi-Dynamics Processor operates as a compressor/limiter, gate, downward expander, ducker or slave. Manual attack and release controls are made dynamically sensitive by active integraters. Packaged in a ½ rack space, use two units for true stereo.

Price: Available upon request

Model 528 Voice Processor is useful for both mic and line level signals. Contains a preamp, de-esser, compressor/limiter, downward expander, parametric EQ and +48 V phantom power for condenser mics.

Price: $649.00

UREI-JBL PROFESSIONAL

LA-4 Compressor/Limiter features long-life LED Optical Attenuator, smooth RMS action, selectable compression ratios, true standard volume indicator (VU), input overload indicator and stereo coupling.

Price: $620.00

7710 LImiter/Compressor features new "Smart Slope" compression characteristics, both peak and average gain reduction, user control of threshold, peak/average blend, attack and release times and ratio controls. Has "automatic preset" pushbutton and comes in single rack space unit.

Price: $495.00

YAMAHA PROFESSIONAL AUDIO —See our ad on page 10-11
GC2020BII 2-Channel Compressor/Limiter can be used in dual mono or stereo modes. Features variable compression, attack, release time, plus variable threshold on expander/gate section. Packaged in a single rack space unit. Weight: 6.6 lbs.

Price: Available upon request

NOISE REDUCTION

DOLBY LABORATORIES INC.
Model 400 Series contains up to 8 channels of switchable Dolby SR and A-type noise reduction. Features auto-alignment, flexible channels assignments, record/play changeover and Autocompare circuitry. Consists of Model 400 controller and up to 4 Model 401 2 channel housings, each occupying a single rack space.

Price: Available upon request

MT Series contains up to 24 channels of switchable Dolby SR and A-type noise reduction. Features auto-alignment, flexible channels assignments, record/play changeover and Autocompare circuitry. Consists of a card frame to accomodate up to 24 Cat.N. 445 modules (8.75 x 19 x 19 in.) and Ps4 power supply/controller.

Price: Available upon request

XPSR Series contains up to 24 channels of Dolby SR (Cat No. 431) modules. Has individual channel bypass, uncal controls and auto compare circuitry. Interchange with Cat.No 331 modules for Dolby A applications.

Price: Available upon request

Model 363 is a two-channel 1-U high frame system switchable between Dolby SR and Dolby A. Has auto record/play changeover, built-in Dolby noise and Dolby tone generators, and auto-compare circuitry. Available in three versions: switchable SR/A, SR only and A only.

Price: Available upon request

Model SDU4 is designed for reference monitoring of Dolby Stereo or Dolby Surround program material. Accepts two-track matrix encoded signal as its input and generates 4 output signals: left, center, right, surround.

Price: Available upon request

ROCKTRON CORP.
HUSH IIX single-ended noise reduction features variable filter cut-off, full bandwidth metering,line/instrument switch and variable threshold in a ½ rack size. Provides 50 dB of effective noise reduction.

Price: $209.00

HUSH IIBX single-ended noise reduction features variable filter cut-off, full bandwidth metering, line/instrument switch, variable threshold, slow or fast expansion release in a single rack space. Provides up to 50 dB of effective noise reduction.

Price: $329.00

HUSH IICX single-ended noise reduction features variable filter cut-off, full bandwidth metering, line/instrument switch, and stereo master in a single rack space. Provides 50 dB of effective noise reduction.

Price: $439.00

HUSH 2000 single-ended noise reduction features variable filter cut-off, filter release, filter sensitivity, stereo link, gating, fully adjustable expander, balanced XLRs and full-bandwidth metering.

Price: $799.00

Model 180A is an eight-channel encode/decode noise reduction system features no switching for record or playback, unbalanced in/out, –10 dB or +4 dB operation. Gives up to 40 dB of effective noise reduction without side effects. Housed in a single rack-space unit.

Price: $799.00

SYMETRIX
511A single-ended noise reduction eliminates noise from any audio source without encoding. Combination of fast dynamic filter and downward expander yields up to 30 dB S/N improvement. Stereo/two-channel switch and subsonic filter included.

Price: $629.00

ADDRESSES

Alesis Corporation
3630 Holdredge Avenue
Los Angeles, CA 90016

Altec Lansing Corporation
10500 West Reno Avenue
Oklahoma City, OK 73128

Aphex Systems Ltd.
11068 Randall Street
Sun Valley, CA 91352

ARX Systems
P.O. Box 842
Silverado, CA 92676-0842

Ashly Audio Inc.
100 Fernwood Avenue
Rochester, NY 14621

DOD Electronics Corporation
5639 South Riley Street
Salt Lake City, UT 84102

Dolby Laboratories Inc.
100 Potrero Avenue
San Francisco, CA 94103-4813

Furman Sound Inc.
30 Rich Street
Greenbrae, CA 94904

LT Sound
7980 LT Parkway
Lithonia, GA 30058

Orban, a Division of AKG Acoustics, Inc.
645 Bryant Street
San Francisco, CA 94107

Peavey Audio Media Research
711 A Street
Meridian, MS 39301

Quad-Eight
225 Parkside Drive
San Fernando, CA 91340

Rane Professional Audio
10802 47th Avenue West
Everett, WA 98204

Rocktron/RSP Technologies
2870 Technology Drive
Rochester Hills, MI 48309

Sonic Image Ltd.
1100 Wheaton Oaks Court
Wheaton, IL 60187-3051

Symetrix Inc.
4211 24th Avenue West
Seattle, WA 98199

UREI-JBL Professional
8500 Balboa Boulevard
Northridge, CA 91329

Yamaha Professional Audio Division
6600 Orangethorpe Avenue
Buena Park, CA 90620

Classified

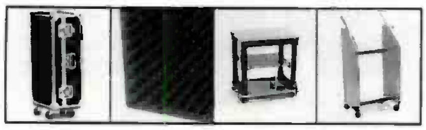
FOR SALE

AUDITRONICS CONSOLE, 24 IN 8 OUT, $6000.00 or Best Offer. **Contact Bill Bailey 404-934-7756** For Complete Information.

<u>How To Produce Great Radio Commercials</u>, by **Brian Battles**. This unique four-cassette package contains essential tips and advice that teach you how to set up your own lucrative advertising production business, and the presentation showcases sample spots using many of the techniques described in this column, and much more.

To order your copy, please send a check or money order for $99.95 (includes domestic shipping) to: Porkpie Productions, P.O. Box 176, Colchester, CT 06415-0176.

Custom Cases Acoustic Foam Custom Stands Studio Furniture



Request Catalogue 800-343-1433 , 516-563-0633
Island Cases • 1120-20 Lincoln Ave. • Holbrook, NY 11741

MAGNETIC RECORDING HEADS RELAP/REPLACEMENT for Audio, Video, Time Code, Duplication. 30 years of head design experience. **IEM**, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.

WANTED

RADIO TRANCRIPTION DISCS: Any size, speed. Drama, comedy, music, variety, adventure, soaps, childrens, AFRS, big band remotes, library services. **Kiner-db, Box 724, Redmond, WA 98073-0724.**

Closing date is the first of the second month preceding the date of issue.

Rates are $1.00 per word with a $25.00 minimum. Boxed ads are $40.00 per column inch. db Box Numbers are $8.50 additional for wording "Department XX" plus $1.50 additional for postage and handling.

Quantity discounts are: 3X – 15%; 6X – 30%.

<u>ALL CLASSIFIEDS MUST BE PREPAID</u>

Send copy to:

db, The Sound Engineering Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.

People, Places... & Happenings

• **Guy Costa announced the formation of Quadim Corporation** and the completion of its first phase of equipment installation for real time audio cassette duplication. Costa comments, "In creating Quadim I wanted to offer the finest in technical facilities, services and the ultimate in quality so we designed a computer system that not only controls the machines but also allows us to individually test each cassette for azimuth, frequency response, distortion, wow & flutter and noise by processing it with the Audio Precision, System One. The computer automates and enhances the operators listening function by randomly selecting the playback of the three-head Nakamichi MR1 machines while intermixing the master source material. Additionally, the computer will perform the routine maintenance checks of the entire duplication line and prevent any machine that does not meet specifications from recording until replaced and retested." Standard services include loading of custom length AGFA 649, TDK SA & METAL tapes into Shape Mark 10 shells, laser printed labels and J-cards. Shrink wrapping, mastering and order fulfillment services are also available.

• **illbruck USA**, the North American unit of SONEX acoustical foam products, has acquired **Northern Sound**, Minneapolis. The acquisition gives illbruck expanded capabilities including: sound/noise level consulting services, a line of electronic noise exposure analyzing and sound level metering instrumentation, and a diverse line of special-application acoustical modification products. All acoustical products have been organized under the **illbruck Acoustical Products Group**, which will move from **illbruck USA's** headquarters to **Northern Sound's** offices at 5155 East River Road N.E., Minneapolis. **Al Perez**, founder of **Northern Sound**, has joined illbruck to provide consulting services.

• **Crown International** President **Clyde Moore** recently announced that **Gerry Barclay** will take charge of the Elkhart, Indiana-based company's marketing services post effective immediately. Barclay, who has been given the official title of Marketing Services Coordinator, will be responsible for: developing advertising campaigns, promotions, artist endorsements, and sales literature for Crown's electronic products line. Barclay will also serve as a liaison between Crown and their advertising and public relations agencies.

• **Altec Lansing Corporation**, President **Dave Merrey** has appointed **Charles "Chuck" Lange** to the position of Vice President, Sales and Marketing. Lange received his BS in Marketing from the University of Wisconsin and is a member of the American Marketing Association. Prior to joining Altec Lansing, Lange was Vice President, Sales and Merchandising for **The Bolen Corporation** in Milwaukee.

• **Chris Stone** has departed as President of the **L.A. Record Plant**, as announced by **Joe Kiener**, Executive Vice President North America of **Chrysalis Group plc**. In December 1987, Chrysalis purchased 50 percent of Record Plant plus one share, and has agreed to exercise its option to become sole owner of the studio operation, including Record Plant's remote and rental divisions. The parting was amicable as "I have nothing but good feelings about our arrangement," remarked Stone. Stone and partner **Gary Kellgren** founded the original Record Plant in New York City, 1967. The Los Angeles Record Plant opened in 1969 on Third Street, with the new facilities on Sycamore Street completed in January, 1986.

• **Sony Professional Audio Division** has named **Robert Ott** to the newly created position of National Business Manager of Microphone Products, as announced by **Osamu Tamura**, Vice President. Prior to joining Sony, Mr. Ott spent 12 years with **Shure Brothers, Inc.**, where he held a variety of positions in microphone and electronics sales.

• **Paul V. Hugo** has been appointed national sales manager of **Gauss Loudspeakers**. Mr. Hugo joins **Gauss Loudspeakers** from **Audio Reinforcement Technologies Inc.**, Miami, Florida, where he was a co-founder and Vice President. Also, **Joe O'Connor** has been promoted to customer service manager, tape duplicators, at **Gauss**. O'Connor, who has been at **Gauss** for eight years, had been customer service engineer, tape duplicators, before being named manager.

• **John Sacchetti**, former chief engineer at **Westlake Audio** in Los Angeles, has joined the **Symetrix** Digital Processing Recorder DPR100 development team.

• **SPARS** recently hosted a meeting at **Paisley Park Studios** in Chanhassen, Minnesota, in an ongoing effort to unite audio professionals in all markets. **Dick Trump, SPARS** Regional VP/Treasurer and President of **Triad Productions**, Des Moines, was the keynote speaker. **Shirley Kaye, SPARS** Executive Director, addressed the meeting with a pledge: "We are here to show that **SPARS** operates on the local level as well as the national...We hope to build even stronger regional identity like we see here for the benefit of the entire audio industry."

• **Ted Pine** has been named to the position of Marketing Manager at **New England Digital Corporation**, according to **Franklin B. Sullivan**, Vice President of Sales and Marketing. Pine assumes responsibility for the development and management of marketing, advertising, direct mail, and public relations programs. Pine joined New England Digital in 1987, bringing with him an extensive background in music recording, composition, and studio management.