THE SOUND ENGINEERING MAGAZINE

DECEMBER 1981
$1.95

db

When he was 16, Humberto moved to the U.S. from Chile, where several of his relatives were successful singers. He worked on an assembly line for a while, before wandering into MGM Studios. A year later, when an engineer got sick before a major session, Humberto was the only one around who could get the job done. He's been getting the job done ever since for an incredible variety of people, from Debbie Boone to Alice Cooper, as well as Frank Sinatra, Sammy Davis Jr., Steve Lawrence, Tony Bennett, Shaun Cassidy, The Osmonds, David Bowie, Denise Williams, Gladys Knight, Bill Champlin, Lee Ritenour, Hall and Oates, Leo Sayer, The Average White Band and Bernie Taupin, whose album he produced.

ON RECORD BUYERS

"When you make hits, you have to think hits—14, 18, young. The people have to be realistic. How many albums is a 27-year-old guy going to buy, as opposed to a 15-year-old? I mean, you go to a record store. Maybe a 16-year-old is going to buy four albums. A 23-year-old is going to buy one or two—he's very picky. He might buy very specific groups that he likes. He might follow critics. When you make records, you have to think kids. Those are the guys who buy the records."

ON RETAKES

"I hate perfect records. You cut the basic track, the vocals, and then the producer goes all the way back again. He starts replacing the drums. And then he replaces the bass, because the bass doesn't feel quite right. And then he starts doing the keyboards again. So that by the time he's finished, he's done it all over again. If it's not right, I understand. Let's do it all over again. But when you start patching things that already have the specific feel in there—that 'something' that has already been printed—you can hear all the human things that are all there for the first time—I don't want to be a part of that. I have been part of one of those and it just drove me crazy."

ON NOISE REDUCTION

"I don't use any noise reduction. I never use it, either when I'm doing tracks or when I'm doing final mixes. They really affect the music. They affect sound in general. To me, the punch is all gone. The drums sound different. The vocals sound different. The keyboards sound different. I can hear those things and it really bothers me, so I don't want to be a part of it."

ON TAPE

"Since I started with MGM, we always used Scotch. Only once, I've experienced a different brand of tape. And I was very disappointed. And I had a serious problem. It got so bad, like in the middle of the mixes, the tape started giving up —heavy drop-out in places. And then the tape started peeling. Not on the outside. It was giving up on the inside. I mean, I was doing a mix, and halfway through the song, the whole top end disappeared, like someone threw a blanket on top of the speaker. So we mixed about halfway through the album. We mixed in sections. We cleaned the heads all over the place. We did the introduction. Clean the heads again. We don't want to take chances. I wouldn't do a project with any other tape besides the 250. I have done the past 20 albums, the past 30 albums all on Scotch. It gives me what I want, and what I want is a real clean taping, punchy bottom end, very little hiss, almost none. You have to try things in order to know if you're doing the right thing. If you don't try, you'll never know. And I have tried, and the results have been different."

SCOTCH 250
WHEN YOU LISTEN FOR A LIVING.

Publisher
Larry Zide

Editor
John M. Woram

Associate Editor
Mark B. Waldstein

Advertising Production & Layout
Kathy Lee

Classified Advertising
Carol Vitelli

Book Sales
Lydia Calogrides

Circulation Manager
Eloise Beach

Art Director
Bob Laurie

Graphics
K&S Graphics

Typography
Spartan Phototype Co.

sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75237
First Continental Bank Bldg.
5801 Marvin D. Love Freeway
Suite 303
(214) 941-4461

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East
Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive
Suite 210
(713) 988-5005

Los Angeles Area
Glendale, California 91204
424 West Colorado Street
Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
(503) 640-2011

San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
(415) 653-2122

Sagamore Publishing Co., Inc.
New York
Plainview, NY 11803
1120 Old Country Rd.
(516) 433-6530

Next Month

• We start the new year with the second part of our feature on Sigma Sound. In part II, David Holmes brings us up-to-date on Sigma's new Sphere console.

And, we'll have John Borwick's report on the latest analog and digital technology from Studer, Jack Gordon will conclude his description of the Gordon Headroom Meter, we'll have our 1981 cumulative index, an AES convention report and more, in **db—The Sound Engineering Magazine.**

db

THE SOUND ENGINEERING MAGAZINE
DECEMBER 1981 VOLUME 15, NO 12

DEPARTMENTS

FEATURES

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application 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 Canada) in U.S. funds. Single copies are $1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

db Letters

MULTI-TRACK SIGNAL-TO-NOISE

TO THE EDITOR:

I must find fault with the footnote to Mr. Rettinger's otherwise fine article on control room acoustics (Oct. '81).

A common error when discussing S/N ratio in summed multi-track systems is to combine noise without considering the signal; a very important part of signal-to-noise ratio! Were all 20 channels carrying the identical signal, the S/N ratio would actually improve 13 dB. In real life, the 20 tracks will contain non-coherent signals, and many will be added in at less than unity gain, resulting in a S/N approaching that of the single channel.

JOHN H. ROBERTS
President
Phoenix Audio Laboratory, Inc.

db replies:

Well, yes and no. It is common practice to measure the noise without considering the actual signal. And, many tracks will indeed be combined at less than unity gain, as Mr. Roberts points out. But, we've never heard a multi-track tape whose signal-to-noise ratio comes close to that of a single channel. If it did, there would probably be a lot fewer sales of noise reduction systems.

Index of Advertisers

db Calendar

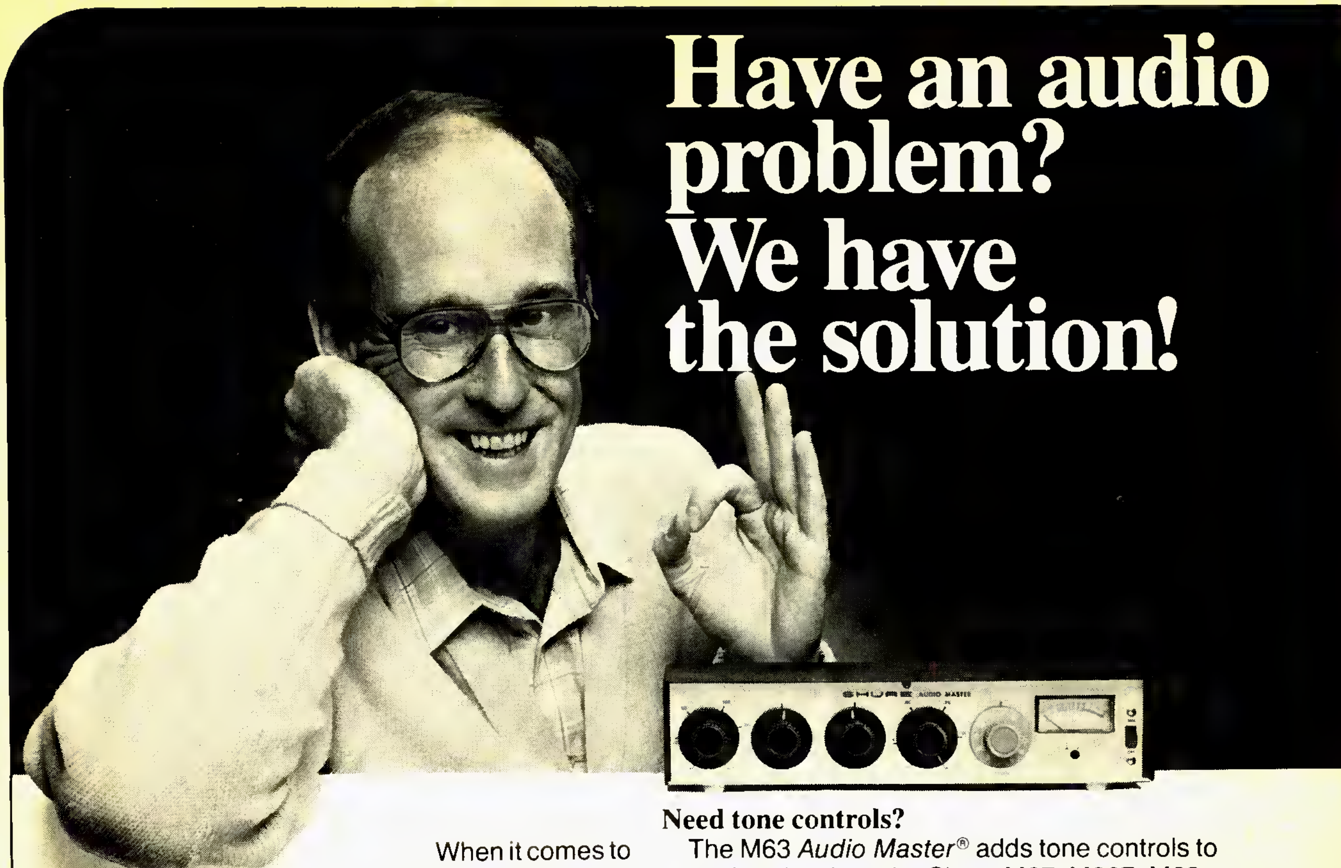
FEBRUARY

5-6 **16th Annual Television Conference of the Society of Motion Picture and Television Engineers.** Opryland Hotel, Nashville, TN. For more information contact: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583. Tel: (914) 472-6606.

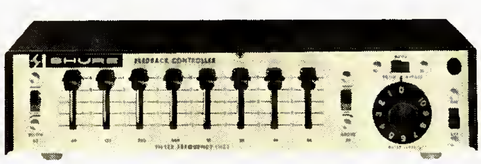
6 **Pay TV: Challenges and Opportunities for the Creative Community.** Sponsored by The Arts, UCLA Extension, Room 2160E Dickson Hall Auditorium, UCLA. For more information contact: Barbara Marcus, UCLA Extension, P.O. Box 24901, Los Angeles, CA 90024. Tel: (213) 825-9064.

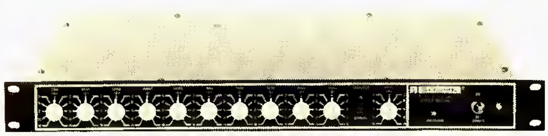
24-26 **Electro-optics, fiber optics and lasers for non-electrical engineers.** Given by George Washington University, Washington, D.C. For more information contact: Director, Continuing Engineering Education, George Washington University, Washington, D.C. 20052. Tel: (800) 424-9773.

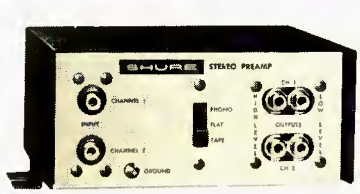
Have an audio problem? We have the solution!



When it comes to solving your audio expansion, equalizing and control problems, Shure offers a full line of equipment designed to handle your toughest assignments. Here's a sampling:







Need tone controls?

The M63 *Audio Master*® adds tone controls to any mixer (such as the Shure M67, M267, M68, M268, SE30). A must for broadcast operations over phone lines.

- Bass and treble tone controls
- Adjustable high- and low-pass filters

Got feedback problems?

The M610 Feedback Controller is specially designed to control feedback in PA systems. Also excellent in post-production rooms for eliminating unwanted background noise from broadcasts, tapes.

- Octave filters centered at 10 key frequencies

Want "audio sweetening"?

The SR107 Audio Equalizer provides "audio sweetening" in post-production rooms for audio and video tapes, and room equalization for hotel, restaurant, church public address systems—perfect where rack space is at a premium.

Trying to boost a low-amplitude signal?

The M64 Stereo Preamplifier boosts low-level signals in broadcast, recording, editing, and signal routing applications.

- Accepts phono or tape input
- Low cost, very versatile

Write for our FREE Circuitry Catalog to get the full story on Shure's lineup of professional circuitry equipment.

SHURE®

The Sound of the Professionals®

Shure Brothers Inc., Dept. 67
222 Hartrey Ave., Evanston, IL 60204
In Canada: A. C. Simmonds & Sons Limited

Manufacturer of high fidelity components, microphones, loudspeakers, sound systems and related circuitry.

Circle 12 on Reader Service Card

To The Editor:

In response to the continuing controversy over stress associated wth listening to digitized audio; perhaps Dr. Diamond is correct in noting a significantly higher stress level in patients after listening to digitized audio. (See "Human Stress Provoked By Digitalized Recordings" in our January issue—Ed.) I believe, however, that he and many others are approaching the possibilities for this occurrence's happening from entirely the wrong direction. I have now spent several hours' time listening to the digital tape machine manufactured by Mitsubishi, and definitely do perceive in myself a feeling of "stress." That "stress" is the same feeling I associate with any sort of live performance; something which is lost in conventional analog recordings. My point, then, is that people seem to be assuming that analog recording is "correct" and digital recording has added something which causes stress. I feel it's the other way around; digital recording is accurate, and something is lost in analog recording. I have never been to a concert where I haven't felt some degree of "stress," and analog recording just hasn't captured that sensation of being in a concert situation. Hurrah for digital!

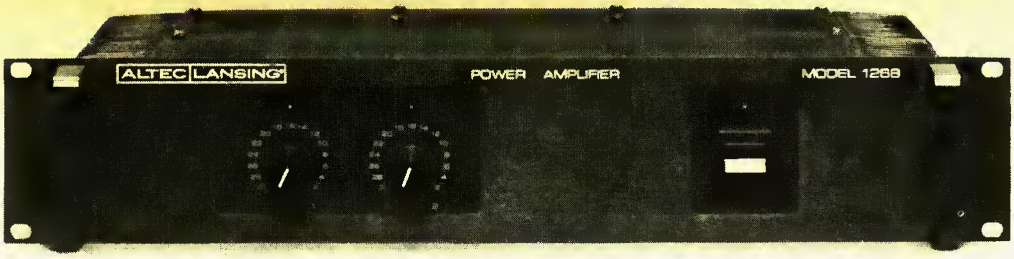
Steven J. Hebrock
Caribou Ranch

db replies:

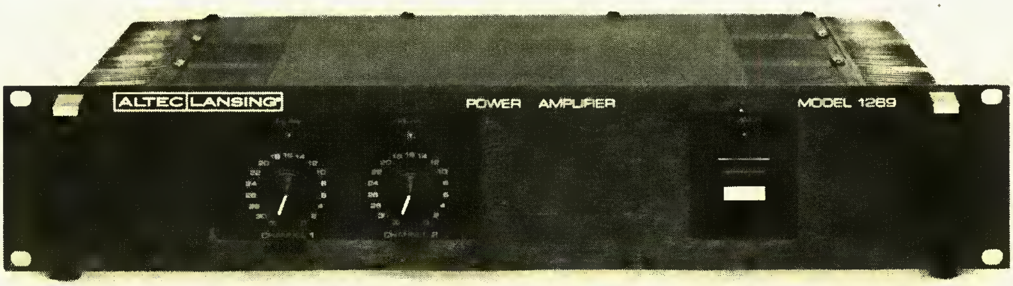
In our April Special Report on "The Digital Controversy, Continued" we expressed similar thoughts: "...what if we eventually discovered that—at last!—digital technology is capable of faithfully capturing all *the joys of music, including its tension and stress?" In the meantime, our research project at the University of Miami has gotten underway, and we hope to have a little something more to say before very much longer.*

About The Cover

• National Video Center & Recording Studios Inc. (NYC) Studio 3 Control Room. Equipment package set up for a music mix for a video assignment: Clockwise starting at 6:00; an MCI Console and 24 track machine; the Q-Lock Synchronized Mixing System (seated on Console) which has enslaved the 24-track recorder with an Ampex 1-in. video or a JVC ¾-in. machine to an ATR 100 four track audio machine. In the foreground of the photo is the JVC ¾-in. machine. Photo courtesy of Robert Wolsch Designs.



THREE NEW AMPLIFIERS.



NINE CONFIGURATIONS.

ONE NAME—ALTEC LANSING.

If you could believe everything you read, there would seem to be a number of companies out there with "positively the best amplifier on the market today." But, when it came to developing a totally new, cost-effective power source for heavy-duty professional use, Altec Lansing was the name people watched for.

And now, the waiting's over.

Presenting the new 1268, 1269 and 1270 dual channel Professional Power Amplifiers from Altec Lansing. Combining advanced computer protection circuitry, major component innovations and computer design techniques, these new units are simply the best performing, most ruggedly reliable power amplifiers in Altec history. Internal circuitry is laid out clean and tight. Servicing is quick and simple. Construction inside and out is state-of-the-art, with features to prove it.

THD, even in the bridge mode with power levels in the 1270 exceeding 800 watts, is less than 0.05% (20 Hz to 20 kHz)—less than that at lower output levels. Power configurations include independent, stereo or mono capabilities. And the 1200's are equipped with efficient, instantaneous VI limiters, restricting output to 400 VA ±45° phase shift, for real-world load-handling most competing amplifiers just can't duplicate. Then there's extensive computer monitoring for protection of amplifier and load, including computerized peak/error comparison of channel input/output signals. All in packages requiring only 3½ inches of rack space for the 1268 and 1269, and 5¼ inches for the 1270.

And the high tech cost-efficiency doesn't stop there. Let us tell you more about the finest professional amplifiers we ever put our name on. Contact Altec Lansing, 1515 South Manchester, Anaheim, CA 92803 (714) 774-2900.

	Power Output per channel		
Model	8 ohms	4 ohms	8 ohms (bridge)
1268	60	120	240
1269	120	200	400
1270	220	400	800

ALTEC LANSING

Circle 28 on Reader Service Card

LEN FELDMAN

db Sound With Images

Professional Sound Using Home Video Recorders

• More and more professional and semi-pro audio people are getting involved in video and turning their portable and home ½-inch VCRs into money-making equipment. Of course, the ¾-inch U-Matic VCR format developed for industrial and professional use can do a much better job of recording, both in terms of picture quality and sound quality, but the cost of setting up a U-Matic based audio/video operation is far greater than the cost of working with a home-type Beta or VHS system. All things considered, picture quality of both Beta and VHS machines is remarkably good—especially at the faster tape speeds (SP, or "standard play" for VHS machines, and Beta II for Betamax types). For most machines that I have measured, video frequency response (which determines picture detail or resolution) extends out to 2.0 MHz and beyond—which is not much poorer than the resolution capabilities of most color TV sets in the first place.

However, when it comes to sound, remember that even at their fastest speeds, VCRs move the tape along at only 0.79 inches per second in Beta format machines and at 1.31 ips in VHS machines. Both of these speeds are well below the already-slow 1.88 ips (i.e., 1⅞) tape speed employed in stereo cassette recorders. Furthermore, while the tape used in VCRs is ½-inch wide, most of that width is used for the video signal's transcription, with only a very narrow track along an edge of the tape assigned to audio. No wonder then that the very best home-type VCRs have an audio response that doesn't exceed much beyond 10 kHz and signal-to-noise ratios of between 40 and 48 dB or so. Add to this the fact that most VCRs employ ALC (automatic level controls) in their audio circuitry, and you can see (and hear!) what a problem it can be to come up with a decent sound track on a video cassette employing half-inch video tape. Still, there are some steps that the serious and enterprising videographer can take to create a sound track that's worthy of the picture accompanying it. And the big advantage with video (as opposed to film-making) is the fact that sound and picture are always in perfect synchronization if both are recorded in real time.

START WITH THE RIGHT MICS

If you are serious about the sound accompanying your video taping efforts, the first thing to do is ignore the omni-directional mic that comes built into most home video color cameras. This microphone usually does a splendid job recording traffic sounds, aircraft passing overhead and *especially* the "heavy breathing" of the camera operator. Even in a reasonably quiet environment, the built-in camera mic remains at a fairly great distance from the scene being videotaped—and everyone who has dealt with that situation in sound recording is familiar with the kind of echo-laden, overly reverberant sound quality that voices take on when recorded by distant mic placements. In video work, the situation becomes even more incongruous, because more often than not you will be using the zoom "close-up" capabilities of your camera lens. This gives you the dubious advantage of a video close-up of your subject, while the subject's voice sounds like it's off in the distance during playback.

Fortunately, most portable VCRs make provision for an external microphone.

Getting that mic close to the subject now frees you up, so that you can move far away from the scene you are shooting, if such long shots are called for.

In addition to equipping yourself with a suitable assortment of microphone extension cables, *don't* overlook getting a decent set of headphones. While most cameras come equipped with those low-cost single-ear phone devices normally associated with cheap transistor radios of yesteryear, you'll want to use your own phones that can cover both ears and let you concentrate on what the microphone is really picking up. (Don't forget an adaptor plug.)

EVM™ by Electro-Voice
The low-frequency loudspeaker

Electro-Voice EVM™ Series II loudspeakers represent the ultimate in maximum efficiency, low-frequency speaker design. Years of experience, testing and refinement have resulted in a series of loudspeakers that are ideally suited for professional high-level, high-quality musical instrument and sound reinforcement systems.

Series II speakers incorporate many unique and innovative refinements that result in a loudspeaker that combines incredibly high power handling capability, efficiency and mechanical durability. All EVM's are conservatively rated at 200 continuous watts per EIA Standard RS-426A. This procedure is substantially more stringent than the more common continuous or "RMS" sine wave test, because it provides not only a 200-watt long-term stress (heat) but also duplicates mechanically demanding short duration program peaks of up to 800 watts which can destroy speaker cones and suspension parts.

EVM's are the ideal speaker for vented and horn-type enclosures. They are also featured in Electro-Voice's TL line of optimally-vented low-frequency systems. TL enclosure builder's plans are also available for custom construction, and each EVM data sheet contains the Thiele/Small parameters which allow you to predict the large and small signal performance in vented boxes.

For these and other reasons, not the least of which is an unmatched record of reliability, EVM's have been universally accepted by sound consultants, contractors and touring sound companies. When specifying a low-frequency loudspeaker, there really is no other choice. EVM – by Electro-Voice.

Electro-Voice®
a gulton company

600 Cecil Street, Buchanan, Michigan 49107

In Canada:
Electro-Voice, Div. of Gulton Industries (Canada) Ltd., 345 Herbert St., Gananoque, Ontario K7G 2V1.

Circle 33 on Reader Service Card

MIC MIXING AND AUDIO DUBBING

As you become more involved in video production you will certainly want to use multiple microphones for making your video sound track, much as you have always done when recording audio alone. Shure Bros. has recently introduced a relatively low cost general purpose mixer. The model M268, pictured here, can be powered by 120-volt AC line, an external 30-volt DC supply, or an optional external battery pack.

In addition to its four microphone-level inputs—each high- or low-impedance selectable and with simplex power available—this mixer has an aux-level input with its own volume control. The master control regulates total program output, and both microphone and aux-level outputs are provided. User net price of the Shure M268 is $250.00 and it is only one of many available portable mic mixers that would be suitable for use with home VCRs. (The Shure M267 is also available, and was described in last month's "New Products"—Ed.)

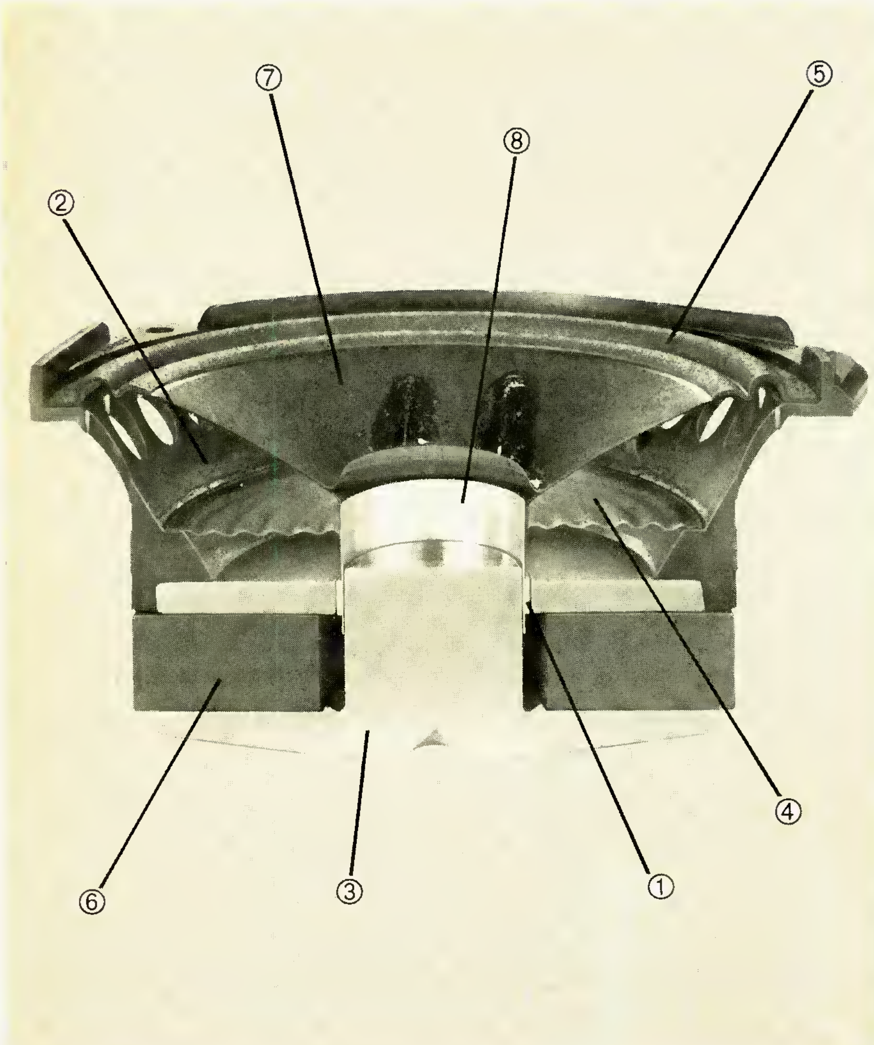
After working with my own VCRs for a few months, I learned that all of the procedures found in the owner's manual regarding audio dubbing should be completely ignored. You see, the makers of these VCRs assume that we (and all of the people who are going to do our voice-overs and music add-ons, fades and segues) are so experienced and professional that all we ever need is one "take." They suggest that we simply plug a mic into the mic jack of the VCR, hit the audio dub button, and substitute our new sounds for the live sound that was originally recorded with the picture. Besides the fact that this procedure obliterates the original sound track (which we often want to include in the new sound mix), getting all the musical and voice pieces to fit together and on cue the first time is something no pro audio person would ever count on. In fact, it probably wouldn't even occur to most of us that we would get it all perfectly on "Take 1."

I've found that a much better technique involves transcribing the existing audio sound track from the video tape onto some form of audio tape so that it can be used as one of the multiple inputs for your final sound mix. With a second audio tape deck you can now combine the new sound sources (from mic mixer, records, tapes, or whatever combination of program sources you wish) to create a new master sound track on the second audio recorder. It is the playback of this final mix that should now be connected to the high-level (line) audio input jack on your VCR (not the external mic input) for final audio dubbing.

To be sure, this system requires care in synchronization, but there are several things that can help you to maintain reasonable synchronization, at least for scenes of relatively short duration. Since you are playing back the master tape on the same machine on which it was made, tape speed should be consistent for the short times required to transcribe a single scene or take. Also, since your final mix also contains the original "live recorded" sound track that is already perfect in terms of lip-synch, you should be able to use that element of the total sound picture to synchronize sights and sounds perfectly. As for the VCR's speed consistency, you don't have to worry about that tape transport system, since it is synchronized electronically to the NTSC standard 30 frames per second of video and will therefore not vary in the speed or pitch of its audio track either. ■

The Bose® 802 Driver.

High technology. High performance.



Efficiency.

Low-impedance voice coil ① uses edgewound aluminum ribbon for maximum conversion efficiency. Nonmagnetic frame ② and one-piece backplate/centerpole ③ reduce flux leakage and wasted fringe field energy.

Accuracy.

Planar spider assembly ④ and foam surround ⑤ provide high linear excursion and low distortion. Syncom® II computer analysis controls total radiated power spectrum of each unit for unprecedented manufacturing consistency.

Power.

12-ounce high-energy Ferrite V magnet ⑥ increases overall sensitivity and power output. Curvilinear cone shape ⑦ maximizes effective piston area.

Reliability.

Glass-reinforced polymer frame ② is impervious to warping. Aluminum bobbin ⑧ and exclusive high-temperature insulation system protect against voice coil deformation and burnout.

The advanced materials and construction of the Bose 802 Driver provide performance unobtainable from speakers of conventional design. And the acoustically coupled array of *eight* drivers in each Bose 802 System delivers smooth response, clarity and bass output that belie its compact size.

For more technical data and a demonstration of the Bose 802 Loudspeaker System, contact your authorized Bose Professional Products Dealer.

BOSE®
Better sound through research.

Covered by patent rights issued and/or pending.
802 speaker design is a trademark of Bose Corporation.

Bose Corporation
Department SE
The Mountain
Framingham, Massachusetts 01701

Please send me a copy of the Bose Professional Products Catalog and a complete dealer list.

Name____________________

Address____________________

City____________________

State__________ Zip__________

Telephone (   )________________

NORMAN H. CROWHURST

db Theory & Practice

Artist or Engineer

• It seems that musicians can hear things at times that audio engineers can't. That can be frustrating. The musician can hear it, but the audio man can't, and therefore assumes that the musician must be imagining it.

Audio people, like other engineers, tend to think they know what they know, and some things are just not capable of contradiction. For instance, Fletcher and Munson conducted tests on human hearing a long time ago, which showed the frequency limits to be 20 to 20,000 Hz, at the extreme. In fact, most people can't hear 20,000 Hz or anywhere near it.

Then you meet a man who says he can hear 27,000 hertz. Impossible!? But you get some equipment capable of reproducing that high, run some test, and find that, in truth, he can hear it, although the rest of us can't. A sort of logic seems to be operating here. I say "sort of," because really it isn't logical, but it seems so. We tend to think, "My hearing is good, I'm not deaf, so if I can't hear it, the sound isn't there."

Although we know there is a threshold of hearing, we don't fully comprehend that it means: sounds (acoustic vibrations if you prefer) not intense enough to reach your personal threshold, will not be audible to you. When you think about it, that is perfectly logical. But if you can't hear it, your natural instinct wants to deny that there is any sound.

We have said before that, way back in the early days of audio, measurements were made to determine what the minimum distortion a person could hear. And in those days, the answer, supported apparently by empirical evidence, was that distortion below 5 percent equivalent harmonic was inaudible. And when that was the latest information, it was accepted as fact.

If someone claimed that he could hear 2 percent distortion, his claim was rejected as ridiculous. Today things have changed. We know that amounts of distortion much lower than that can be heard. But how could those early experimenters have been so far off?

What were they listening for? What could they listen for? To determine when they could hear a difference, between sound with and without the distortion. If you can't hear the difference, then it was not detectable, right? Apparently not, as we have since found.

The skilled auto mechanic listens to your car engine, and tells you that you've got a bearing "going." He may even tell you which one. But when you listen, you can't hear how your engine sounds any different from the way it has always

TRUTH REVEALED!

TOA VMS SERIES ENDURES RACK TORTURE FOR YEARS.

Introducing painless P.A. and sound reinforcement: the Variable Multi-function System of professional and commercial sound components—VMS. No matter what kind of torture a sound contractor devises, a TOA VMS modular system can take it.

VMS is a wide range of engineered sound components that are readily adaptable to an equally wide range of uses. There's over a dozen plug-in choices of signal processing modules that go into a self-powered (A.C. or D.C.) mainframe preamplifier mixer. Divided into separate preamp and line amp sections, this VMS Series 2000 electronics package can be expanded up to three stacked mainframes loaded with audio modules like chime units, signal generators, filters, meters, a complete selection of preamps, and a line amp. From here you go to an electronic crossover, one-third octave graphic equalizer, your choice of several rugged power amplifiers including a bi-amped model, and two versions of convenient monitoring packages. All in a standard 19" rack.

Although your custom-tailored TOA VMS rack components may never see the light of day, you can whip-up a powerful speaker system with a combination of high and low frequency drivers that will tell the truth in just about any size environment. With your choice of VMS electronics and transducer systems, you now have many ways to make a quality sound system speak— with crystalline clarity.

VMS can spell quick relief, too, with deliveries right out of stock. So you can get into and out of a job right away, without delays and supplier uncertainties. And last, but not least, we must confess that the prices for VMS components are competitive with anybody else in the business. With VMS you can rest assured that we thoroughly intend to inflict pain to the old-fashioned ways of doing things (like we've been doing for the last three years).

To tell the truth, we're going to be around for many more years to come—enduring the rack tortures of demanding, high-quality sound system applications, taking the worst possible punishments a sound pro can sentence us to.

VMS. By TOA. Call us now for all the gory details.

Crafted in Japan
Proven in the States.

TOA Sound & Communications

TOA Electronics, Inc.
1023 Grandview Drive
South San Francisco, CA 94080
(415) 588-2538 Telex: 331-332

Circle 13 on Reader Service Card

sounded. He knows what to listen for; you don't. He "sorts out" that mass of sounds that you just take as being "the engine." His ear is trained as a diagnostic tool.

Audio people's ears are trained too, and so are musician's. But their trainings are different. An audio man listens to his equipment. When he has instruments to measure it, he checks frequency response, dynamic range, distortion of various kinds. If he is experienced, he will tell you that the response is from here to there, within so many dB, and the distortion is less-than-so-much.

A musician listens to his music; he listens to how well he can hear what was going on at the input end in terms of someone playing musical instruments, not in terms of what frequencies are present. The audio man hears frequency; the musician hears pitch and timbre. And while one is translatable into the other, they are essentially different.

Relativity theory tells us, and has data to prove, that energy and matter are translatable. The atom bomb proved that matter can be converted into energy. Maybe it isn't quite so hard to convert an instrument's pitch and timbre into a real-time frequency analysis, but nevertheless it is a conversion. And according to what our vocation is, we train our ears to work for us accordingly.

To a musician, his musical instrument is an extension of himself. The instrument and the musician become one, so that while playing, he doesn't know where he ends and the instrument begins. His faculties "feel" those sounds out into the air, and he is not conscious of the detailed mechanism by which it all happens. He just "thinks" the music, and his bodily functions, are so coordinated and trained that the music "happens" in complete accord with his thought.

For some reason, if something doesn't work as it should, he becomes conscious that something is wrong. But when that happens, he's not thinking about frequency content and distortion, as would his friend the audio man. What they hear is different. Assume that some element in the amplifier "caves in," doesn't continue working as it should. The audio man, versed in electronics, quickly recognizes the symptoms—he's heard it happen before.

But to the musician, something is "getting in the way" of his music. Maybe it's garbling it with spurious components that mess up the sound. In this connection, one thing has always amazed me, and it gives cause for thought.

I well remember musicians at the "top of their form," who listened to performances on the old acoustic phonograph, and could better criticize the performance (not the recording) that way, than with the newer electrical transcription that has universally replaced it today. To me, the full frequency range of electrical recording was always infinitely clearer

Are you tired of spending 6 digit figures every year for new recording hardware, because you don't have . . .

updatability?

UPDATABILITY is a unique word. So unique that you won't find it in any dictionary.

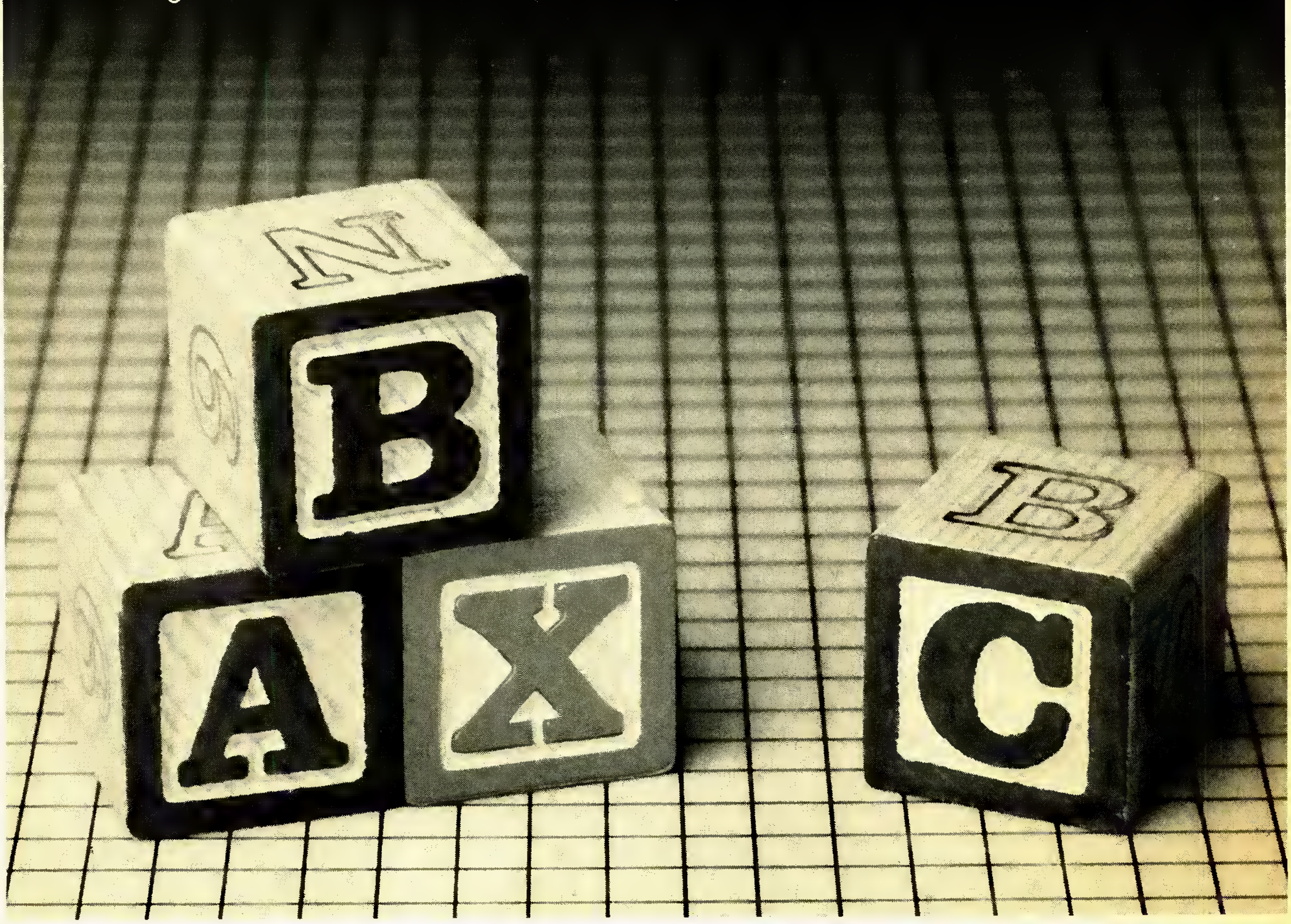
UPDATABILITY is the I.C.C. 3000 Modular Audio Control Console

The I.C.C. 3000 Audio Control Console is functionally superior to anything on the market. Each function has been separated into modular building blocks. This allows for superior sound now and in the future. Technological developments are inevitable. As they occur you simply plug them in. Yes, the I.C.C. 3000 Audio Control Console may be the last console you **will ever** buy!

Hearing is believing!

I.C.C. has put together a library of recording demonstration tapes to exhibit the sonic quality of **our technology.*** These tapes will be played at the A.E.S. convention daily. Should your firm not experience the I.C.C. 3000 Audio Control Console first hand, we will loan you a two track fifteen-inch-per second original master.

Contact I.C.C. for availability. After all, 'hearing is believing'!



Call or write:

I.C.C.
P.O. Box 862
Provo, Utah 84603
(801) 377-9044

*Patents Pending

Circle 26 on Reader Service Card

than the old acoustic phonograph. I began to realize that musicians and audio people hear different things.

To me, the scratches on that old acoustically-recorded disc almost obscured the music. The dynamic range was about 10 dB! The electical recording of those days wasn't up to today's standards, but it was certainly a lot better than the old mechanical-acoustical type—at least to my ears. Yet some of these highly respected musicians rejected the new, in favor of the old: they could hear the music better on the old! I asked myself, "How could that be?"

I learned a long while ago to accept what someone else says, and not to think that he's off-base just because I do not see things in the same way. If he says he can pick up nuances on the acoustic phonograph that he can't hear on early electrical recordings, I ought to believe him: he knows more about music than I do! Maybe I can't explain why, and it doesn't seem logical, but there must be something about it, to explain his experience.

This is a useful viewpoint to adopt. Had I applied the sort of logic that says because I can't hear it, it isn't there, I would have said the auto mechanic could not hear a bearing going bad, just because I couldn't. And on that score, I know better: I've seen bearings taken down that proved the auto mechanic right.

So the thing to do when something along these lines happens is to make a mental note that here is something you don't understand. As far as I know, all modern musicians use electronic recording as the tool with which to analyze performances: none still insist the acoustic phonograph is better. But then, if you compare some of the earlier electrical recordings, with those made today, there is a big difference, too.

Could it be that there was something that the acoustic phonograph did, with regard to "interpreting" certain essential aspects of the musical sound, that the early electrical recordings failed to do? That's something to think about, isn't it? It sets you thinking along the way things developed.

Back in those early days, everything was single channel: monophonic. Today we have stereo. And that conversion was also quite an evolution. We remember days when monophonic was better quality—i.e. lower distortion and better frequency response—than stereo, so some were predicting that stereo was a "fad" that would never last. But the same was said about color, both in motion pictures and later in television. But color is here to stay and, while the early color was inferior to the best black and white of those days, that is no longer true: when you lose color today, you lose some of the picture value.

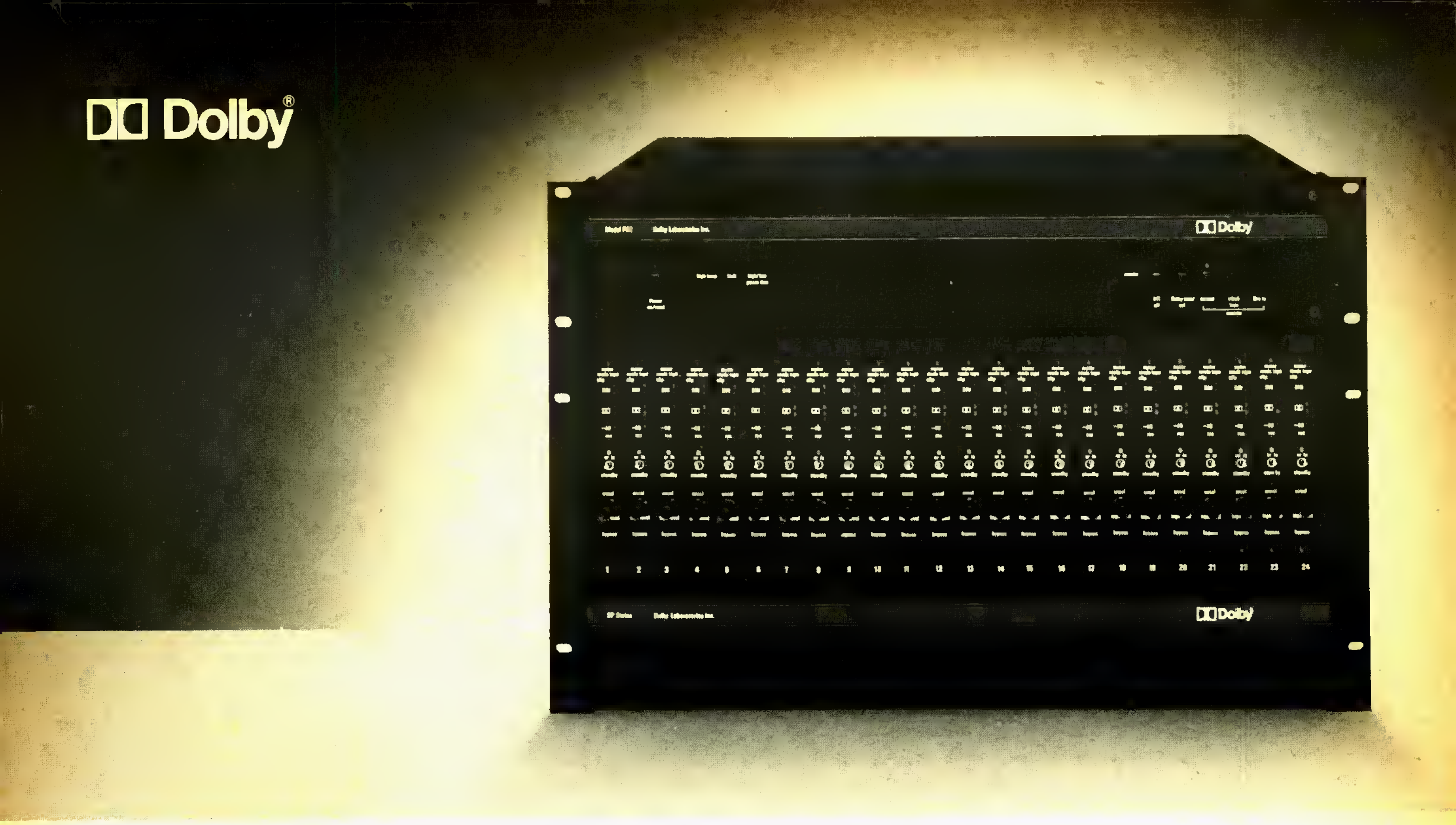
So what is it that stereo adds? Does stereo add something that audio engineers and musicians perceive differently also? Perhaps this was more obvious in the early days of stereo, too. The engineers wanted to demonstrate separation. Because the early stereo quality was inferior to mono quality of the time, the superiority that is immediately obvious today did not show. So they resorted to the so-called "ping pong" effect. Music was specially scored for that purpose.

Dance bands had been doing it, so the trend was to get that kind of music recorded on stereo: the trumpets would play a phrase of music, and the woodwinds would answer from the other side of the stage. The only difference necessary for stereo was to make sure they were as far apart as possible, to get the "separation" on playback. Then the difference between mono and stereo was that with mono, you could tell the brass from the woodwind by the quality of the instruments, while with stereo, it was easier to tell by the fact that the sound came out of a different loudspeaker!

Engineers have gone on working on quality, and now stereo is at least as good as mono ever was. So stereo no longer requires material scored for ping pong effect. Separation means not only spatial separation—the fact that different instruments come from different apparent locations in space—it also means identity separation; the fact that using multiple channels enables you to separate the sounds from different instruments in your head, regardless of where they seem to be placed.

Conductors have always grouped instruments in an orchestra, presumably, to help both them and the musicians to separate their performances more effectively. But that relates to the days when instruments all generated their sounds by an acoustical-mechanical process: vibrating strings, air column in pipes, and so forth. The age of the synthesizer had not arrived.

That led to the big question when the synthesizer eventually did come along—"Where do we put it?"—not so much in the physical sense, as where would it fit into the musical family, sound-wise. Well, it has made it—or perhaps it would be more accurate to say, it is making it, for I am sure that we have a lot to see yet, in what electronic sources of musical sound can contribute. ■



New from Dolby Laboratories:

The Dolby® SP Series Multi-track noise reduction unit

Dolby noise reduction is an integral part of professional multi-track recording practice in music, radio and TV broadcasting, and film studios throughout the world. A new noise reduction unit, the Dolby SP Series, has been developed for these and other applications, and provides up to 24 tracks of Dolby A-type noise reduction in only 12¼" of rack space. The SP Series' combination of compact size, ease of operation, and new features makes it ideal for equipping new recording facilities and upgrading existing ones.

For further information on the SP Series and other professional noise reduction equipment, contact Dolby Laboratories.

Highlights of the Dolby SP Series:

- Up to 24 tracks in only 12¼" of rack space, including power supply.
- Dolby A-type noise reduction characteristics utilizing standard Dolby Cat. No. 22 modules.
- Separate regulated power supply unit with electronically-controlled output protection.
- Low-noise fan cooling.
- LED display for each track permits accurate Dolby level calibration (within ±0.1 dB if desired) by matching intensity of LED pairs; further LEDs indicate the presence of signals and clipping, and assist alignment with high-level reference tapes.
- Front-panel "UNCAL" control for each track permits rapid resetting of Dolby level for playback and punch-in on nonstandard-level tapes, then instant restoration of preferred preset studio Dolby level without recalibration.
- User-selectable option of "hard" or electronically-buffered bypass of individual tracks and of all tracks simultaneously.
- Snap-fit connectors on rear panel for rapid disconnection and reconnection.
- Balanced and floating input stages.
- Output stages drive either single-ended or balanced 600-ohm loads at levels up to +28 dB (19.5 V) before clipping.
- Ultra-low-distortion input and output amplifiers.
- Remote ground-sensing output configuration minimizes hum pickup when driving single-ended loads.
- Discrete FET switching for reliable, noise-free routing of audio signals.

Dolby Laboratories, 731 Sansome Street, San Francisco, California 94111, Telephone 415-392-0300, Telex 34409. 346 Clapham Road, London SW9, Telephone 01-720-1111, Telex 919109. "Dolby" and the double-D symbol are trademarks of Dolby Laboratories Licensing Corporation. S81/3621

BARRY BLESSER

db Digital Audio

• The digital domain has many buzz words and appears quite complex. Many of us have become familiar with the technology without completely understanding it. So, in this and the next few articles, we will focus on trying to develop a basic foundation of digital language in order to show that the concepts are really very simple.

Each of the basic elements of the digital system is so very simple that it can be explained in a few words. The complexity comes because the complete digital system has such a very large number of these elements combined. Thus, the first major task is to develop a language to describe such systems. Please forget your knowledge of the buzz words since they are often better for obscuring rather than illuminating. Let us begin.

IN THE BEGINNING

There are only two basic types of digital circuits: combinatorial (no memory) and sequential (memory). Combinatorial logic derives from *combining*, hence we need a language for this activity.

We start with the definition of the signal. This itself presents a problem since there are two kinds of notations: logical and electrical. Logical is a case of a Name being True or False. Electrical is a case of a wire being High (2.5 volts or more) or Low (0.8 volts or less). H (high) does not necessarily mean True, unless we define it to be so, and then we call it *positive* logic. The matter of a Name being True when High (or when Low) may be illustrated with an example.

Consider a light switch on a wall, with Up and Down corresponding to the physical states which are equivalent to electrical H or L. The logical *name* which we give to the signal may be either LIGHT or DARK. One can talk of the room as being in a state of lightness or darkness. The name DARK is True when the switch is Down, and the name LIGHT is True when the switch is Up. This is a linguistic problem which has confused digital engineers for two decades. Recently, work at the international standards level has tried to resolve some of the confusion in a systematic manner. Now, after the logical name there must be a notation which states the physical condition when that logical name is True. In our example, the wiring in the room would be represented by LIGHT-U or DARK-D. This reads; LIGHT is True when the switch is Up, or, DARK is True when the switch is Down.

If the switch was *physically* rotated, the names would read LIGHT-D or DARK-U. When we wish to think in terms of darkness we can do so regardless of the way that the switch is wired. The reason that this is important is that a given device, such as a NAND gate, can function as either AND or OR logic. The same physical device can have different logical functions! Because of the need to make complex circuits easier to describe, it would be nice if the notation could contain the logical operation and still be a unique characterization of the device.

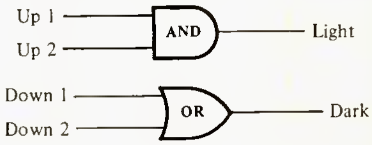
Notice that if one were to see a signal coming from a switch which was represented by LIGHT-D, we would know to think logically in terms of lightness, and we would also know the way that the switch was physically installed.

Only the most recent documentation by more-enlightened engineers contains this full notation. Older documentation requires the reader to mentally do the above exercise, probably while trying to recall a footnote from a previous page which says that the guest bedroom and the back porch have their switches wired upside down. This is easy to forget and it is much better to have the information contained in the symbol.

LOGIC

Before going to real digital logic, let's consider some house-wiring logic in which a local light switch and a master switch both control the lights in one room. Is this wiring an OR or an AND function? It's either one, depending on how we wish to look at it. One could say: if Switch 1 AND Switch 2 are Up, then the room is Light; however, we could also say: if Switch 1 OR Switch 2 is Down, then the room is Dark. To give a logical name to a physical device requires us to define the logic symbols *first*. In these two examples we have not changed the wiring in the house, yet the logical function which describes the relationship between the switches and the light may have two different names. Traditionally, most logic designers give names to create what has been called *positive* logic. The term is actually used in logic data books to describe the devices, because otherwise the device function cannot be stated. However, complex designs lead to complexities which come about from keeping track of uncomfortable names. When a given piece of logic controls darkness, we may prefer to think in terms of the variable DARKNESS rather than of NOT-LIGHT. Therefore, there may be two types of symbols to describe the same logic device.

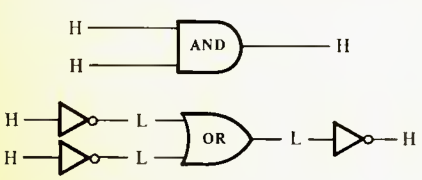
Our house wiring would be represented on a digital diagram as either of the following two cases:



The first symbol says that *both* switches need to be Up to create light; the second says that *either* switch being Down creates darkness. The problem is now one of making sure that the function relating the name and the physical device is unambiguous. Otherwise, we would never know how to relate them. In the past, digital engineers selected names to correspond to the devices using an implied positive logic. This means that for an AND gate, both inputs need to be True to give a True output. We could only represent the first example with the house wiring and we would have to think in terms of switches being Up or NOT-Up and the room being Light or NOT-Light. With so many negations and inversions, it is quite easy to make mistakes. Hence, we should use a notation which tells us what condition to expect when a variable is true.

The industry is beginning to convert to a standard which will help ease this linguistic problem. A signal name is given a suffix to indicate the condition when it is true. The suffix is H or L with electrical logic representing a high voltage or a low voltage. Notice that voltage is physical, not symbolic.

To continue our example, assume that the switches generated logic-type signals corresponding to 5 volts (high) or 0 volts (low) and that the room is illuminated when the logical input to the light is H (high). This is a physical statement, not a logical statement. The logical statement could be represented as the following:

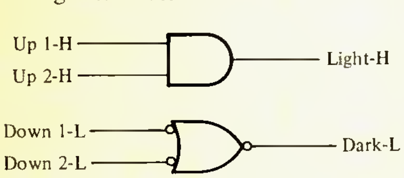


(The small circles are a compact way of showing the inverters.) Both of these symbols illustrate the same functions, but we can select the one corresponding to the way we think. The symbol (ignoring inversions) operates on the name such as: Up AND Up gives Light; alternatively Down OR Down gives Dark.

The same electrical device, of course, may have different symbols. For example, the 7400 NAND gate and the 7402 NOR gate may each be represented in two ways, depending on the logic:

The first example, being positive logic, is very similar to the previous example. The second example is more interesting, since we have modified it to use an OR symbol, yet the electrical relationship between input and output is the same. To get an H output from it, the OR must produce an L; for the OR to produce an L, both of its inputs must be L, which means that the two signals must be high. These two symbols systems both describe the same physical hardware.

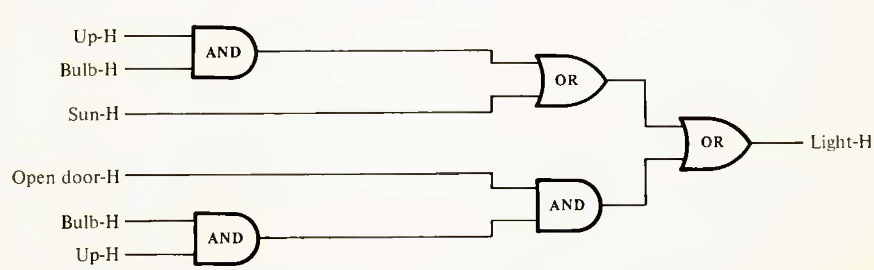
Similarly, the name Up-H and Down-L for the switches both describe the same physical properties of the switch. It is only a case of thinking. This leads us to the final compact notation for the house wiring seen here:



ADVANCED COMBINATIONS

All combinatorial logic circuits are made up of these kinds of elements. We can show that the 7400 NAND can be used to create all device characteristics since it can be used as an inverter also. Inversion-plus-NAND can be used to create OR, NOR, XOR (exclusive OR[1]), etc.

To revert back to our house analogy, consider a more complex environment, in which the state of illumination in a room includes the possibility of the sun, an open door, etc. The logic diagram below says that there will be light in the room if the lamp switch is Up AND there is a bulb, OR there is sunlight, OR there is an open door to the next room, AND that room has a bulb in a lamp, AND the lamp switch is Up. We represent this by the following logic diagram:

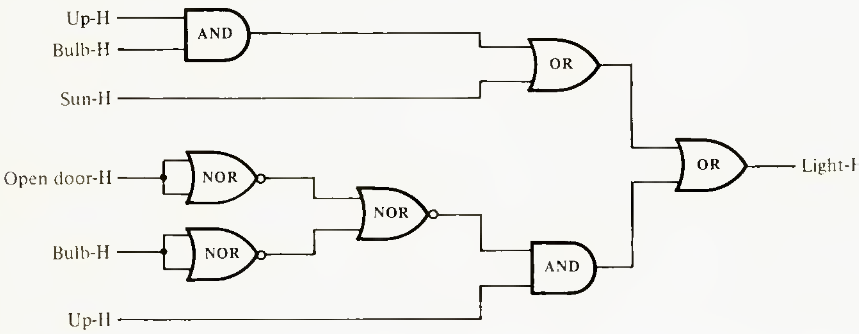


All combinatorial logic is composed in this fashion. The analysis proceeds from input to output and, although tedious, one can determine the state of all the outputs just by combining the various signal names with the logic functions. Such a system has no memory and the output is always determined by the present states of the inputs. Past history makes no difference: Combinatorial and no-memory are equivalent terms.

The design of very complex combinatorial logic is not always straightforward if one wants to minimize the number of gates or devices.

There have been some formal techniques developed and chip designers use computers for a large number of variables. However, the modern digital designer is rarely faced with a very complex combinatorial problem. The use of ROMs (read-only memories) is often a better and simpler solution. Also, the use of PAL (programmable array logic) makes the task easier since one is not concerned with optimizing the number of gates. The fact that there are many digital combinatorial circuits that have the same input-output relationships is no longer very interesting. Sometimes, there will be a few spare gates in a design and it is better to use these rather than add a new circuit to implement a function.

In the previous example, consider a case in which we are short one AND gate, but have three extra NOR gates. The circuit could be modified to be the following:



This is a different circuit, but the relationships are the same. As a black box, one cannot tell the difference. By combining the Open door and Bulb signals, we create a difficulty in thinking since there is no logical reason to think of doors and bulbs. In the previous diagram, the door signal was combined with the light from the lamp, which makes semantic sense. Nevertheless, both circuits will work in an identical fashion. The problem only comes if one has a very large system with hundreds of input and output variables.

There is more than a difference of scale when one talks of 4 input variables compared to 75. Playing chess in the end game with only 4 pieces on the board is much simpler than playing chess with 22 pieces on the board.

We should add one note of warning. When combinatorial logic is combined with sequential logic, the speed at which information can propagate through the combinatorial logic does play a role. For example, the signal Open door needed to go through two levels of gates to get to the output in the first version but 4 levels in the second. Hence, there will be a longer delay time. ■

[1]OR means A OR B OR both. XOR means exclusively A OR B but NOT both. One and only one input must be true for a true output.

db Editorial

ONCE AGAIN, we've survived another four days of wandering the corridors of the Waldorf-Astoria hotel here in Fun City in search of the latest in pro audio hardware. The Audio Engineering Society's 70th convention is at last over, our feet are beginning to recover, and if all goes well, we should have a full report (on the convention, not the feet) by next month.

Needless the say, the ubiquitous micro-processor was discovered lurking behind just about every new front panel in the place, giving small comfort to those of us who are still trying to decide if it's OK to use transistors.

The micro-processor-based electronic music system is gaining in popularity, and the many on display drew SRO crowds throughout the show. One was even interfaced with a graphics plotter, so that once you're satisfied with your composition, you can dash off a hard copy for the benefit of any human-type musicians who may be involved.

As it happens, we ran into one of those human-types, an old friend of ours who was having a bad time of it, trying to decide whether he was delighted or frightened by the goings-on. He wondered what will happen to musicians who are replaced by drum synthesizers. And what about all that lost overtime? (He'd already figured out that you don't even have to pay union scale to a VCA.) On the other hand, he was intrigued by the sounds of software. But on the *other* hand...

We realized our friend needed some consoling and we suggested a hasty retreat to a nearby watering-hole, where we could escape the crowds, drown some olives, and think the whole thing through.

Our friend was concerned: will the studio musician wind up playing second fiddle to a chip? Will a new generation of switched-on programmers wind up doing all the film scoring by 1984? Could we lend him some money towards a new disk drive he needed?

We offered to pay for the next round instead, and assured him that some of the best digital audio we'd heard at the show was coming from the big black box in the Bösendorfer room, and it wasn't even plugged in. Just try *that* on your disk drive!

Our friend thought it was unlikely that anyone would ever come up with a hardware/software package to replace the symphony orchestra (although he was thinking about trying to do so after the show). Of course, he'd never had a symphony orchestra in his studio (20 x 30, with an isolation booth). He did do a string overdub now and then (the carpet rolls up) and he wondered if we knew anyone who had come up with a good viola sound. (There's never enough money for violas.) With some good viola software on hand, he would feel better about going to the bank for a synthesizer loan. With a synthesizer, he knew he could attract more groups (and budget-conscious producers) into his studio, and they could all forget about those expensive string overdubs.

We asked what would happen to all those studio musicians who would be out of work once he got his string software (and a few other sounds) worked out. He accused us of stepping on his lines, and reminded us that we were supposed to be worrying about drum synthesizers. Didn't you use to play drums yourself, we asked? That's got nothing to do with it, he assured us, and anxiously looked for the waitress.

Maybe some of the musicians' locals will eventually get the message, and try to make the job of hiring a studio orchestra a little less unpleasant. No, we're not suggesting sweat shop recording sessions, or even playing for less than scale. We are suggesting a more common-sense approach to clock-watching though. We've all seen many situations in which the session is successfully driven into overtime. Once again, the battle of "getting some OT" is won. Once again, the war against unemployment is a little closer to being lost.

Enter the synthesizer. It takes no coffee breaks, and doesn't slow down as the big hand approaches 12. It doesn't make mistakes deliberately. Does it sound like a symphony orchestra? Only after the fourth martini. Before the first one, it doesn't even sound like one string player.

Certainly, it will never replace all live musicians. Just as certainly, it will replace some of them. Of course, us engineering types have nothing to fear. No one has built an automated mixdown system that corrects balances by itself. Our friend looked up from his olive. "You've just given me a great idea," he said. "Why, with the right software, I could cut my engineering budget in half!"

We quickly asked the waitress for separate checks and saw our friend to his car. Hopefully, he'll drive off a bridge on the way home. It would serve him right, for having such a terrible thought! ■

Circle 23 on Reader Service Card

JOHN BORWICK

A Right Royal Recording System

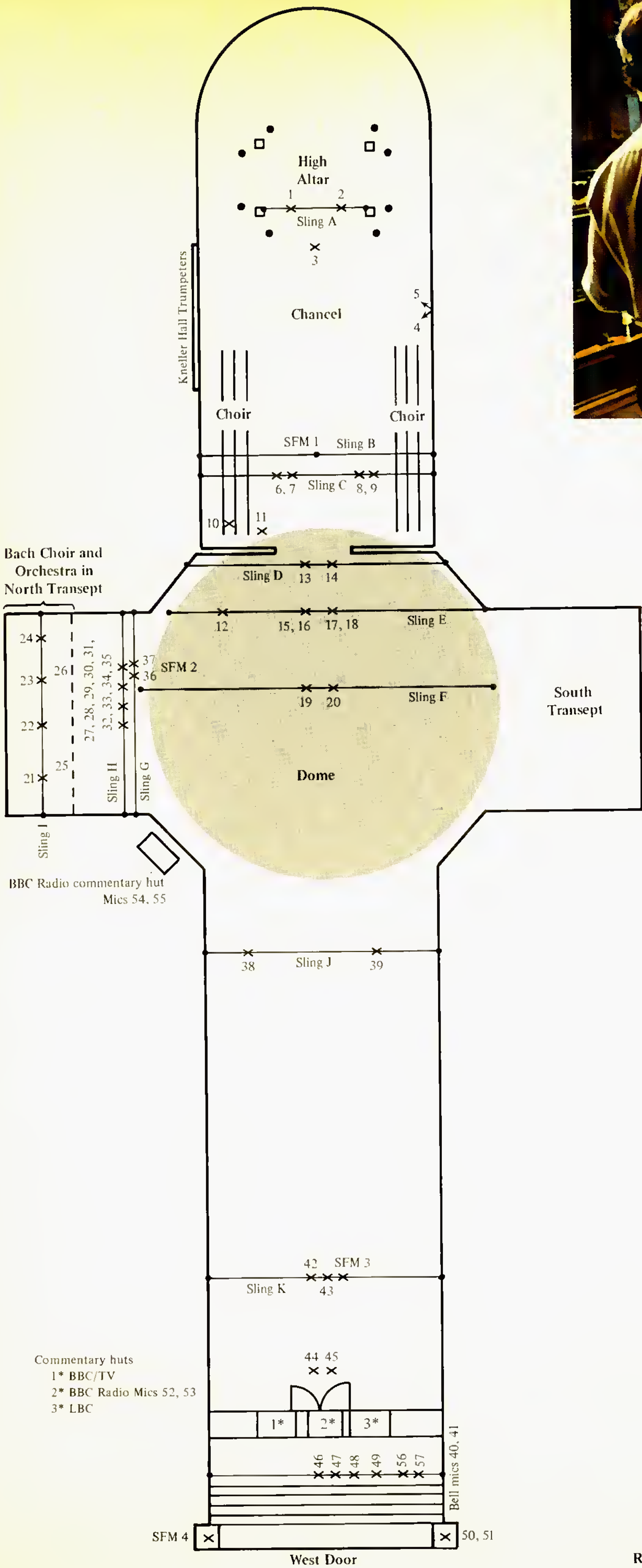
John Borwick takes us behind the scenes at Britain's "Wedding of the century."

BBC cameraman captures part of the procession as it leaves Buckingham Palace on its way to the Cathedral. The coach with Queen Elizabeth and the Duke of Edinburgh is just leaving.

An estimated world-wide audience of 500 million television viewers watched the wedding of Prince Charles and Lady Diana Spencer on July 29th, and many more listened on radio, making the total live audience around one-eighth of the population of this planet. Yet anyone who was foolish enough to miss the show live can console themselves with the video and audio discs and tapes which were rushed into the shops in record time, and became immediate best-sellers.

So that we can concentrate on the audio engineering aspects of this mammoth production, let us get the basic television statistics out of the way first. The BBC (British Broadcasting Corporation) used 44 cameras linked to a Colour Mobile Central Control Room parked alongside St. Paul's Cathedral. The centre portion of this special vehicle has expanding sides to provide a decent production area in which up to 42 monitor screens can be viewed from a comfortable distance. The camera disposition was 12 in the Buckingham Palace area, 6 near

John Borwick is **db***'s European correspondent.*



The BBC's North Choir camera in St. Paul's Cathedral seen here looking along the Chancel towards the marriage dais.

Microphones

- 1, 2 Altar on sling A
- 3 Desk 10' from Altar
- 4, 5 Kneller Hall Trumpets-dropped from triforium
- 6 – 9 Choir
- 10 Officiant-attached to desk
- 11 Lesson-attached to rail
- 12 Organ-on sling E
- 13, 14 Dais-marriage ceremony-on sling D
- 15, 16 Main stereo-on sling E
- 17, 18 Spare stereo-on sling E
- 19, 20 Whispering gallery-state trumpeters-on sling F
- 21 – 24 Bach Choir-sling I
- 25 Harps-on stand
- 26 Tymps-percussion-on stand
- 27 – 29 Woodwind
- 30 Basses-on short rhythm stand
- 31 Solo trumpet
- 32 Solo soprano
- 33 Chamber organ
- 34, 35 Main orchestral stereo mic-on sling H
- 36, 37 Spare orchestral stereo mic-on sling H
- 38, 39 Congregation mics-on sling J
- 40, 41 Bell mics-table stand
- 42, 43 West organ-sling K
- 44, 45 West door-state trumpeters-dropped from triforium
- 46, 47 Fx on pillar-west portico
- 48, 49 FX (standby)-west portico
- 50, 51 Fx-with windshield on SW lantern
- 52, 53 Radio commentators lips-hut 2
- 54, 55 Radio commentators lips-hut in NW quarter dome
- 56, 57 Portico trumpet mics
- Sound field mic 1-dropped vertically from sling B
- Sound field mic 2-on sling G
- Sound field mic 3-on sling K
- Sound field mic 4-in NW lantern

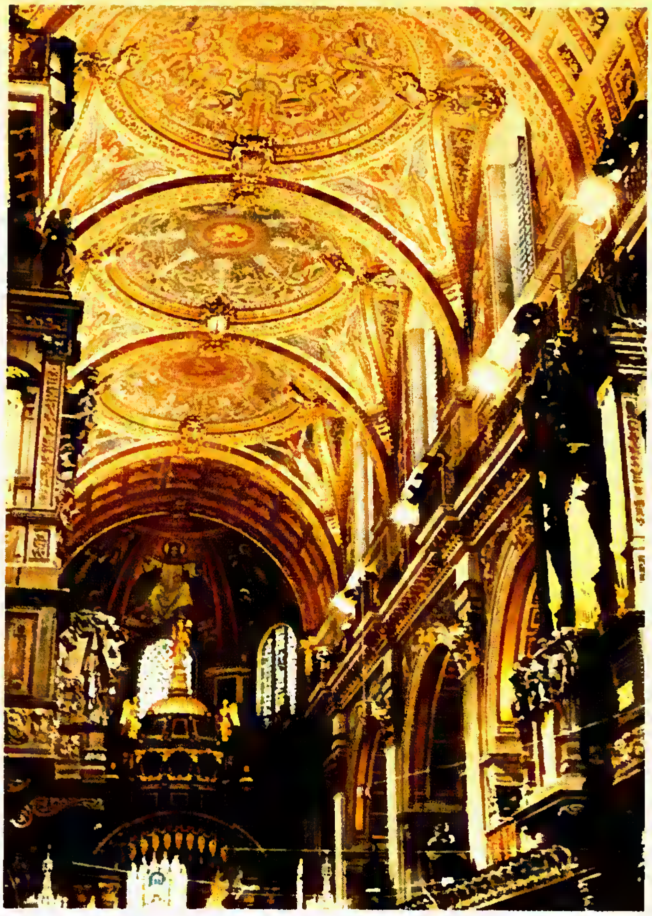
ROYAL WEDDING – ST. PAUL'S CATHEDRAL – JULY 29, 1981

Admiralty Arch (including one on top to provide the aerial views of the processions in the Mall), 4 in the Strand, 3 in the Royal Courts of Justice, 1 on top of New Zealand House, 12 in St. Paul's Cathedral and 6 in the area outside. A mixture of landlines and SHF radio links was used to connect cameras, microphones, talkback, etc. to and from the control centre and its many satellite locations.

The cameras inside the Cathedral were concealed behind false partitions decorated to blend with their surroundings. Many extra lights were rigged and various colour filters were placed over certain windows to achieve the desired colour temperature.

The BBC was responsible for providing all facilities for foreign commentators. Therefore the complete broadcast picture as directed by producer Mike Lumley in the Central Vehicle, and relayed to BBC Television Centre for transmission, was split to 40 separate monitor booths. The sound portion of the programme was sent as an optional "clean-feed," i.e., it consisted of the final mix of music and effects microphones but without the BBC commentator's voice. Each foreign broadcaster could therefore produce his own voice-over. The BBC television programme also went to the European Broadcasting Union in Brussels for distribution to countries throughout Europe, and to three Intelsat satellites supplying the rest of the world.

Twenty-two video tape recorders were kept busy at Television Centre, including sets of three each in three editing suites which generated "edited highlights" programmes from lunchtime onwards. The new Palantype instantaneous sub-titling system enabled hard-of-hearing viewers on BBC to follow the live commentary describing the processions, while pre-recorded subtitles were used for the Cathedral service.



View from the Chancel of the ceiling at St. Paul's. Some of the microphone slings for the Cathedral Choir are shown at the bottom of the photograph.

BBC engineers sling some of the microphones in St. Paul's Cathedral, directed by Harold Kutscheraurer (centre).

INDEPENDENT TELEVISION TOO

While the BBC provided the largest coverage and assumed responsibility for world-wide distribution (free of charge), they were certainly not alone. All three main American networks had installed programme teams and superstar commentators and the Independent Television Authority, competing fiercely with the BBC for Britain's viewing audience, had a full-scale telecast team in operation. They used 40 main cameras and a dozen mobile electronic ones. To spearhead their attack they also had a camera and commentator floating high over London aboard a Goodyear blimp (helicopters having been ruled out as too noisy on such an important occasion).

AUDIO IN THE CATHEDRAL

The sound of the wedding ceremony in St. Paul's Cathedral was entirely the responsibility of BBC Radio engineers, the signals then being relayed by them to BBC radio and TV, independent radio and TV, and an incredible number of other broadcasters at home and abroad.

Sound balance was engineered by BBC Senior Sound Supervisor Harold Kutscherauer, who has been making a specialty of ceremonial occasions for the past 16 years and is very familiar with the acoustic problems of the Cathedral. In fact, as anyone who has attended a service there will know, the built-in sound reinforcement system is exceptionally elaborate and successful so that the entire congregation area provides good clean sound despite the 11 seconds reverberation period and such acoustic oddities as the enormous dome and "Whispering Gallery."

The Tascam 16-Track System.

Chair not included.

All this for the price of a 2-inch recorder.

If you don't need 2-inch compatibility or 24 tracks, you don't have to pay the price for expensive 2-inch hardware.

Instead, you can own the heart of a 16-track studio.

The Tascam Model 15 Mixer. 24-in 8-out 16-track monitor with a comprehensive cue system that can be fed simultaneously by 48 signals.

The Tascam 85-16 Recorder/Reproducer. A 1-inch transport with 16-track integral dbx.*

And the Tascam 35-2B Mastering Recorder. A 1/4-inch 2-track with integral dbx.

With this Tascam 16-track system, you'll get professional sound at an affordable price. And because it's all new equipment, you save even more with tax credit potentials.

Write to us today for the name of your nearest authorized Tascam dealer. He'll show you how a Tascam 16-track system can fit your studio needs and your budget.

TASCAM
TEAC Production Products

*dbx is a registered trademark of dbx inc.

Circle 18 on Reader Service Card

However, producing a satisfactory balance of the large and varied musical programme and every single word of the ceremony—when 1,000 million people are listening—was no simple matter.

Harold Kutscherauer had rigged some 57 microphones and sat at a 64-channel mixer in the Cyrpt producing a straight stereo mix to be sent as a "clean-feed" (without commentaries) to most locations. At a second "mixed-feed" mixer, Peter Hunt added the commentaries for BBC radio, while racks of microphone splitter boxes (producing two buffered outputs from each of the 57 microphones) enabled various users to select some of the microphones direct. BBC television, for example, had run all the microphones out to their control vehicle and did their own sound mixing.

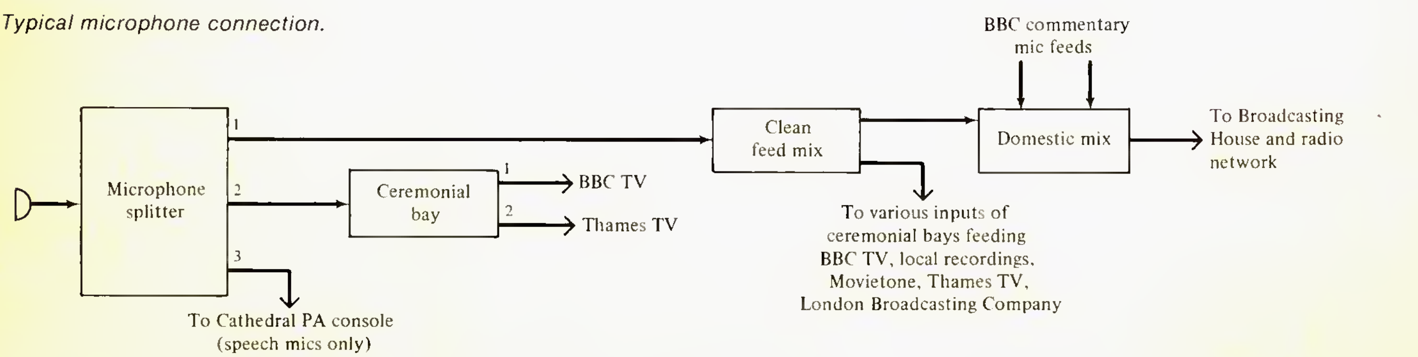
The microphones were mostly AKG and Neumann capacitor types, many of them stereo models set in various coincident-pair configurations, and slings were used where possible. St. Paul's Cathedral—like many frequently used broadcasting/recording venues in Britain—has permanently installed XLR socket terminals and concealed tie-lines. In addition, some microphones, such as those over the ceremonial dais and altar, hung straight down on their own (strengthened) cables from the 70-feet high triforium gallery.

Particular areas to be covered, apart from the dais and altar, included the Chancel (Cathedral Choir and Kneller Hall trumpeters), the Whispering Gallery and West Door (State trumpeters), the North Transept (Orchestra and Bach Choir), the organ pipes in the North-East Quarter Dome and above the West Door, and the Cathedral bells. While a natural-sounding overall balance was the key criterion, a good number of spot microphones was necessary. This helped sound-only listeners who, of course, had no visual clues as to what was happening. It is also standard practice in television sound to give the TV producer the option of slightly (hopefully imperceptibly) raising the level of soloists, etc. as they appear in close-up.

As always on large location assignments, the presence of numerous TV cameras and the associated variable lighting circuits gave the sound engineers extra problems. In this case, there were 7 miles of microphone cable and 10 miles of lighting. So tracking down sources of mains hum and thyristor dimmer sizzles took quite a long time at rehearsal, but the usual techniques of careful grounding and re-routing of the more intransigent cable lengths proved successful.

Harold Kutscherāurer, the BBC's Senior Sound Supervisor, at the controls of the ceremonial (clean feed) mixer in the BBC's Crypt Control Room.

Typical microphone connection.



Mixed feed position housed in the BBC Control Room in the Crypt of St. Paul's. Peter Hunt is at the controls.

RECORDING FOR POSTERITY

Video and audio recording rooms were working at full capacity throughout this exciting day-long event. Apart from the normal edited versions to supply newscasts and magazine programmes at frequent intervals, there was a decision to produce both video and audio cassettes and discs, and get them out on sale to the public in the shortest possible time. The BBC's LP record was cut and rushed through the presses at several disc factories to appear within 24 hours. It had reached Number 27 on the charts three days later, having already sold 100,000 copies. The BBC also supplied the disc to Teledisc, who sold it packaged as a two-disc album along with suitable stirring music, and aimed to achieve 150,000 sales by direct TV-response advertising on Independent Television. The BBC's 90-minute videocassette also sold well. By the end of the week following the Wedding, about 5,000 cassettes had been sold. The programme included the Service, the processions, the balcony appearance, the honeymoon departure and excerpts from the splendid music performed in St. Paul's Cathedral.

At the same time, the BBC made a number of experimental digital and surround-sound recordings. A digital van at St. Paul's housed twin Sony 16-bit PCM recorders. This recorded the clean-feed sound, with a dust-free recording area built to guard against drop-outs. For the same reason, special high quality tape was used here, and at the BBC's digital recording room back in Broadcasting House where a mixed-feed recording was made.

Again for experimental purposes, and following an expression of interest from Buckingham Palace itself, BBC engineers recorded the whole ceremony with Ambisonics UHJ encoding. The balance engineer for this was Bob Harrison, who has considerable experience in surround-sound recording. He used four Calrec Soundfield microphones which, as regular readers of **db** Magazine will know, each comprise a cluster of four capsules producing signals capable of reproducing the original soundfield, with height information if required. These microphones were suspended fairly close to the BBC's main coincident-pair stereo microphones, and of course could be "steered" electronically during the subsequent UHJ encoding to give almost any desired degree of directionality. Bob Harrison judged the best height for each of his microphones at rehearsal by listening in the omnidirectional mode, and he settled for heights of about 25, 30 and 40 feet respectively for the orchestra, choir and congregation microphones. (See "The Sound Field Microphone" and "From England: The Ambisonics System" in our July and August, 1978 isues—Ed.)

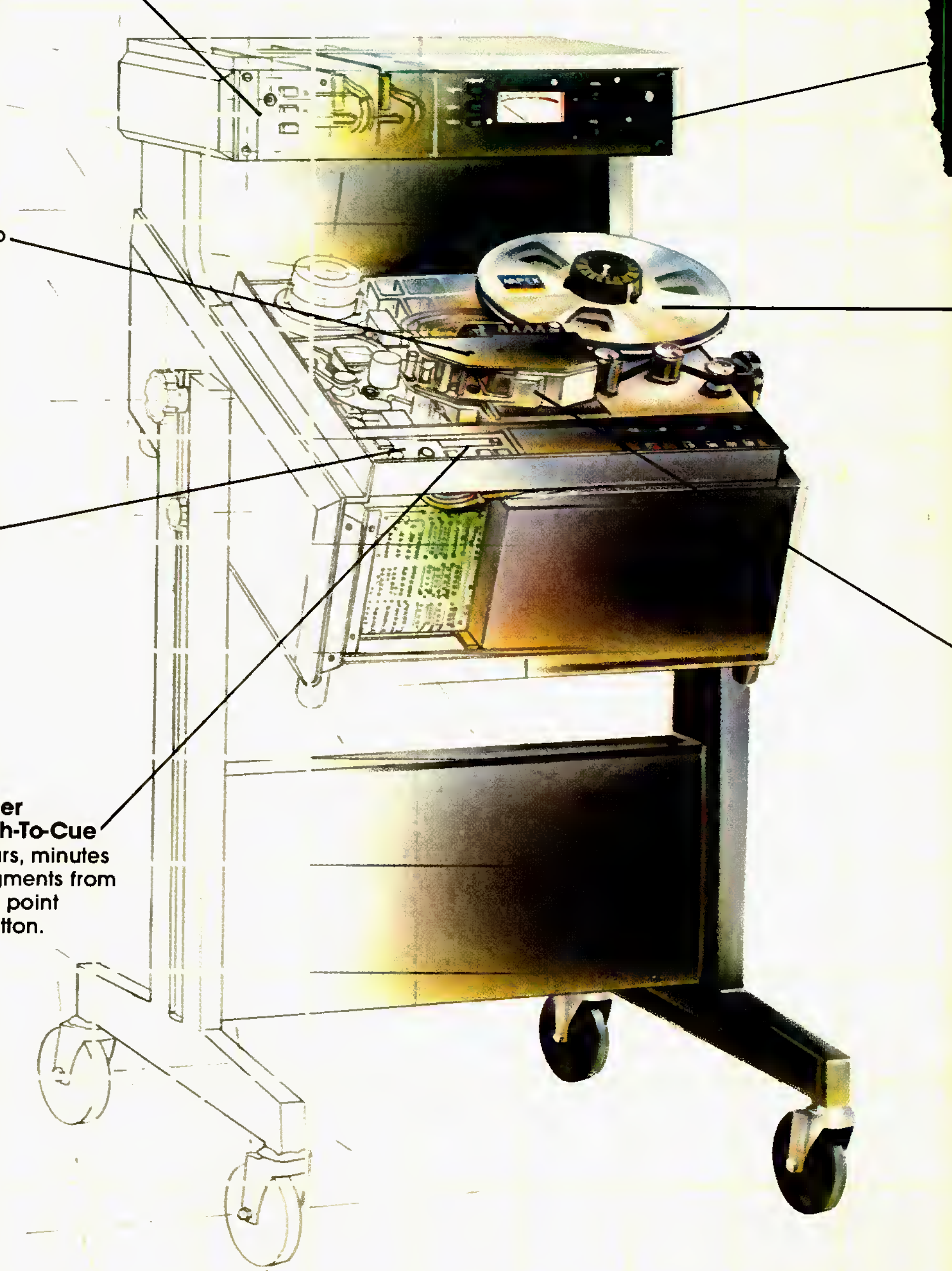
To allow maximum flexibility at the later mixing and editing stage, the 16 signals from the four Soundfield microphones were recorded straight to 24-track Studer M800 tape machines, running at 30 ips with no noise reduction. The other tracks were filled by stereo effects microphones situated just outside the Cathedral and in the Whispering Gallery, the stereo mix from the BBC mixer and an SMPTE time-code track. The costs for making this experimental recording were met by the NRDC (National Research and Development Council), who are backing the Ambisonics system. There were many interesting and antiphonal effects in this unique recording venue, including organ, separate trumpet bands, two separate choirs, congregation singing, crowds outside, the Service itself and the musical programme which featured star operatic soprano Kiri Te Kanawa and an 85-piece orchestra. These should make an ideal showcase for the Ambisonic system, both in the UHJ format and in full periphonic sound, and the Ambisonics team are hoping to persuade a major record company, and the various artists and copyright holders involved, to go ahead with LP disc production.

I should like to acknowledge with gratitude the friendly help I have been given in preparing this report, with a special thank you to John Flewitt of BBC Engineering Information Department who took the photographs, Bob Harrison and IMF Electronics. ■

BROADCAST

Serviceability
Major electronic assemblies are plug-in and easily accessible from the front of the recorder, even when rack mounted.

Quick Change Head Assembly
Converts from one to two to four channels, or back quickly with no mechanical re-alignment.



Three Speeds with Variable Speed Operation
Machines are shipped with three speeds, 7½, 15 and 30 in/sec. Field convertible to 3¾, 7½ and 15 in/sec.

Digital Tape Timer with Single-Point Search-To-Cue
For accurate timing in hours, minutes and seconds. Rehearse segments from exactly the same cue point at the touch of a button.

PRECISION

Switchable NAB/IEC Setup
At the flip of a switch, the recorder converts between NAB and IEC setup, including bias and levels as well as equalization curves.

Microprocessor Control
New tape transport design is under the full control of the microprocessor system, ensuring safe, gentle and foolproof tape handling.

Designed For Editing
Head assembly is wide open for unequaled accessibility. Optional tape cutter and marker available. Dump edit and hands-on-reel editing modes included.

Introducing the Ampex ATR-800. More features than ever before in a broadcast audio recorder.

In a busy broadcast environment, every minute counts. That's why Ampex designed the ATR-800 with saving time in mind. With more standard features than any other recorder in its class, the ATR-800 is the perfect choice for the special audio needs of the broadcast professional. And recording studio engineers? Take note.

The ATR-800 was designed for tape editing. The wide open head assembly gives you fast, accurate tape access. Recessed head gate and transport controls prevent tape snag. And a continuously variable shuttle, under control of the microprocessor, regulates tape speed and direction.

You'll find hands-on-reel and tape dump edit modes included for convenience. The standard cue amplifier will allow monitoring of any or all channels, right at the machine while it's being cued. And with flexible transport controls, you can now mount them either to the left or to the right side of the machine —whichever way you choose.

But the features don't stop there. You get a quick change head assembly, a digital tape timer with single-point search-to-cue, three tape speeds with built-in vari-speed, fader start for remote control from a console, simple service access from the front of the recorder and much, much more. All standard. And with a switchable NAB/IEC setup, the ATR-800 is a true international recorder in every sense of the word.

Look around, no other audio recorder has the number of standard features that meet the needs of the broadcast professional like the Ampex ATR-800. It's shipped for rack mount installation, and it's available in console and pedestal versions as well. Look into the ATR–800. Call your Ampex dealer or write Ampex Corporation, Audio-Video Systems Division, 401 Broadway, Redwood City, CA 94063 (415) 367-2011. Sales, spares and service worldwide.

AMPEX

TOOLS FOR TOMORROW

Ampex Corporation • One of The Signal Companies

Circle 24 on Reader Service Card

LANCE PARKER

Audio Consoles: The Real Problems, The Real Solutions

If you're not happy with the way consoles are made, make one yourself! Here is one man's approach to console design.

WHEN I WAS IN THIRD GRADE, the Junior High School Band came to our school and performed. I can remember that it was an outdoor performance during the spring. To my young ears, their music seemed to be a rich, delicious treat. This experience aroused something inside me that was never to be quieted. Think back for a moment. All of us in the music industry can perhaps remember our first awakening to the power and beauty of music. It's like falling in love; it's like a magnet pulling us in. We become addicted, and only more music with increasingly better quality seems to satisfy.

Throughout Elementary School and Junior High School my love affair with music grew. In my Junior year in High School I made the decision to dedicate my life to the enhancement of recordings. I was determined to produce the finest recordings ever made and Heaven help anything that got in my way.

It soon became obvious to me that my biggest stumbling block was the intimidating, illogical, audio control console. I already knew that I could get quality sound by hooking a microphone up to a preamp, the preamp to an equalizer and the equalizer to a tape track, but to my amazement, the console manufacturers were throwing all kinds of garbage into this simple signal path—buffers, transformers, balancers, FET switches, impedance converters, output drivers—I thought they were nuts.

The addition of all these things was just *one* of the problems. Others included limited—if any—updatability, questionable sound quality, lack of versatility, and high cost. Systems are illogically complex and hard to learn. They have also been hard to customize, difficult to add to or expand, unreliable, difficult and expensive to repair, and made with poor quality parts and materials. Other systems don't have enough visual indicators, and suffer poor patching and routing of signals, lost control settings, coldly sterile appearance, difficult matching to semi-pro or non-standard gear, and overly glorified specifications.

Since my goal was to make the finest recordings that have ever been made and since I was faced with so many of these problems, I had to do something. I looked around for help, but no other designer seemed to appreciate the problems. They were too busy adding chrome to last year's models. Being unwilling to accept defeat, I decided to do something about it myself.

The awesome challenge of addressing each and every one of these problems has taken me ten years of research and development. Many times the frustrations have been so overwhelming that, had I been driven by money alone, I probably would have given up.

So much for the history. Now for the good news—THE SOLUTIONS. It is very important that you keep the term UPDATABILITY firmly in mind as you read further because it will unquestionably take on greater significance as we delve into these solutions. The most important solution to the greatest number of problems came after eight years of comprehensive analysis. *Separation By Function*, henceforth referred to simply as SBF (registered), contains the key.

SBF offers a logical means of solving many of the problems. Let me explain the theory behind SBF through the use of the simple lead pencil as an example. Separation by function means exactly what it says. We separate the pencil into three parts: (1) the lead, (2) the wood, (3) the eraser. The lead has the *function* of leaving a mark when pressed against what one is writing on. The wood has the *function* of supporting the lead and providing a gripping surface. The eraser has the *function* of removing the mark left by the lead. Three parts, three *functions*. By separating the pencil by *function*, a whole new world of versatility opens up to us. We can assign a separate design team to concentrate on perfecting each function, namely: the lead, the wood, the eraser. The only thing one team needs to know about the other is how the parts hook together. These teams don't even have to work for the same company. In fact, many teams from different companies may even compete to see who can make the most perfect part.

Lance Parker is the president of International Consoles Corp., Provo, Utah.

The I.C.C. Console.

UPDATABILITY

Now, let's look at all the new possibilities. Keep in mind that everything that applies to the simple functioning pencil can, in the same sense, be applied to consoles. If a preamp with improved sound quality becomes available, it can be easily inserted into our existing console without the necessity of the redesign or replacement of other console functions. SBF allows us to update *any* function for *any* reason whether it be more versatility, improved quality, simplification, increased reliability, etc. Note that I put UPDATABILITY ahead of sound quality. Like most people, I used to think that sound quality was king, with versatility running a close second. But even if you buy the system with the best sound today, the guy down the street is going to buy something better tomorrow. Updatability is king because state-of-the-art sound (and versatility, and everything else in your system) depends on it, including the eventual size of your bank account.

Next, let's take a closer look at versatility. The problem with versatility is two-fold. As things get more versatile, they tend to get more complex and more expensive. Console manufacturers have been adding more knobs and buttons every year. We used to sit down to operate consoles, but now we have to stand up to reach all of those added knobs and buttons. We even pay extra for the privilege. The added complexity makes it difficult to keep track of everything during a session.

I used to think that the solution to the versatility problem was to keep everything simple with very few controls and limited patching, but this can be like trying to paint a beautiful outdoor scene with only black and orange on your palette. The real solution lies in what I call "Chinese vs. English Alphabet soup."

Picture two boys about twelve years of age sitting at a table. Each has a bowl of alphabet soup in front of him. One of the boys speak Chinese and his bowl is huge. The Chinese alphabet requires many different characters for each thought that must be communicated, hence the huge bowl. The other boy speaks English, and with only twenty-six letters in his alphabet, he requires a much smaller bowl. An elderly friend decides to have some fun with the boys. He offers $200.00 to the boy who can most efficiently spell words with his alphabet soup. He then requests certain words to be correctly spelled. After the first hour, the Chinese-speaking boy is still fishing through his huge bowl, searching for the required characters, while the English-speaking boy is efficiently beginning his sixty-first word.

This analogy points out that it is not necessary to have a huge bowl of soup to have versatility. In other words, we need a *small* set of efficient building blocks from which we can construct *any* desired result. Our audio control console should be designed utilizing basic, efficient, separated building blocks. How interesting! We already discovered that updatability depended upon SBF. Now we discover that the problem of versatility is also solvable by SBF, *if* we can identify the basic, efficient, separate building blocks.

VERSATILITY VERSUS GADGETRY

In designing a pencil, we may need different leads for a variety of writing and drafting applications and we may even need different erasers for removing these various marks. But, we can live without warning buzzers which tell us that the lead has just broken or erasers which come in meaningless, pretty colors.

Now let's apply this to consoles. If something *enhances* the final product that leaves the studio, either by directly affecting the signal quality or indirectly by causing a pleasant experience for those involved in producing and engineering that product, then it belongs in the category of Basic Versatility. On the other hand, if you can say to yourself, "We could have achieved just as good a product *without* the device in question," then it's probably a gadget. This will help to explain why I have such a problem with some computer devices. If I stick the darned thing inside the console along with everything else, SBF tells me I'll end up regretting that I didn't put it in a freestanding cabinet.

MORE PROBLEMS AND SOLUTIONS

Let's go on now to the next problem: high cost. For most of us, high cost becomes problem number one when our current console is out-dated. SBF offers armor against obsolescence because you need only replace the outdated functions without worrying about the rest. The cost of getting into the initial system, which is modular and expandable, can be lower simply because you're buying only what you need now—and along with the knowledge that you can expand your system when you desire and when you can afford it. Keep in mind that a system with basic, efficient, separate building blocks can be less expensive than Chinese Alphabet Soup.

The next problem is that consoles are too illogically complex and hard to learn. The solution to this problem is both easy and difficult. The easy part is that all one has to do is put the controls in order, flowing from top to bottom in each channel. The difficult part lies in the fact that the order changes depending on the type of recording or mixing you're doing. At first, this seemed to me an insurmountable problem, but after experimenting with a multitude of block diagrams, I finally discovered a logical solution. Only *one* device (other than the pre post sends) must move around in the sequential order of things—the Tape Track. The reason all consoles cannot visually move their tape track around is that the bus switches at the top of the channel are the only thing that represent the track position. Obviously, they cannot be moved around in logical order.

The solution is to strategically place LEDs at three points in the channel which illuminate the word TRACK when the track is at that point in the sequential order of the channel. The "lock-step" idea that one must always have Bus 1 tied to the input of Track 1, with a bus switch for each track, *had to be scrubbed* in order for a logical ordering of controls to be made possible. Having the controls in their logical sequence, from top to bottom in the channel, as well as separating the control sections by function, makes the control console *easy to learn*. Now, by having the buses free-floating from the tracks, a whole new world of busing opens up to us. One can use the buses to go *anywhere* in the system simply by having modules that can select any bus's feed at any place in the system. I call these Input Selector Modules. You will be hearing more about these later in this article.

Consoles are not easily customized. Obviously, no console manufacturer can offer you precisely everything that you need and want, but if a manufacturer offers basic, efficient, separate building blocks, you will be able to combine these, much like the ABCs in our alphabet, in any amount and arrangement to satisfy your particular needs; i.e. numbers of tape tracks, effects, microphones, monitors, cue systems, etc. Consoles are difficult to add to or expand, since manufacturers offer their console cabinets in set sizes. This makes sense from a structural standpoint, since cabinets need to be strong. However, you can't sell the small studio a huge cabinet with lots of blanks included just because they might need the expansion space. It's too expensive and expansive.

The solution to this problem is the expandable cabinet. To keep the cabinet simple and structurally strong, I recommend that at least one part of the cabinet system be replaced rather than telescoped or added to when an expandable cabinet design is considered. The part of the cabinet system I refer to is the structural part that spans from the left legs to the right legs and supports the modules' weight. For example, in the I.C.C. 3000 we use ten 2-in. x 2-in. aluminum angles. When the owner wishes to expand his console he merely replaces these angles with longer ones and inserts more motherboard sections and modules. Even the armrest in an expandable cabinet should be modular to allow for future expansion. Expansion is almost unlimited with this approach.

The next three problems: some consoles are unreliable, difficult and expensive to repair, and have poor quality parts and materials. If you buy a big console you had better buy a big repairman.

There is no universal solution to these problems. Every part that's going to fail doesn't always fail during the manufacturers "burn-in" testing procedures. Rather, it fails right in the middle of your biggest recording session. The console operator has two options available to him when this happens. He can either route around the problem or replace the defective parts of the system.

If the system is separated by function into modules, the offending module is easily identified and replaced with a spare. Also, if everything in the system is modular, you can find the offending module through a simple trading process. To avoid problems in the first place, choose LEDs over incandescent lights; double-up on contacts, *all* contacts. Redundant back-ups proved adequate when putting men on the Moon, so it's fairly safe to assume that similar back-up systems within your console will get you through a session. If, after all this, a problem re-occurs, the simple re-design and replacement of the offending module provides a permanent solution. Hopefully, the application of these techniques will make your repairman as much in demand as the Maytag repairman.

The next problem I wish to discuss is the lack of visual indicators. The real problem here is: What am I using on the console? Where am I using it? And how much am I using?

Without visual indicators, we have to keep track of all this in our heads. On a console with several thousand controls, this can be very difficult. The solution to this problem is to have controls which light up when in use, and if these controls are arranged in logical order, all the better. Controls not in use are darkened and therefore mentally bypassed. This answers the questions: What am I using on the console and where am I using it?

Now: how much am I using? We have learned from experience that to get the best sound out of a tape recorder, we must drive the tape as hard as possible without distortion. The same holds true for every active circuit in an audio control console. Each stage must be driven as hard as possible without distortion to get that HOT sound. VU meters which read only the bus outputs are simply not enough. We need to know what the mic preamp and the equalizer are doing, too. This means more meters, and if they are to be placed where they belong in the sequential control order, then they will have to be smaller and probably a different shape than a bar. The challenge to the console designer is to design these indicators in such a way that they blend in tastefully. Otherwise, consoles can begin to look more like they belong in a discotheque.

The next problem concerns itself with cumbersome patching and routing of signals. My solution to this problem has been to use buses on the console to patch and route signals. This is made possible because, as discussed before, the buses are not permanently dedicated to track inputs, but are instead free-floating, and may be addressed to any part of the console system. This is similar to giving a telephone to each device in the system allowing it to call any other device within the system. On the surface this may sound complex, but it really isn't.

The output of one device is routed onto a bus by selecting the bus on a numbered switch. The same number is selected on a switch at the Input of the device which is to receive the signal.

This new method of patching and routing is modular and automatically grows with the switch. It replaces patch cords and, if configured properly, eliminates signal ground, level, and balancing problems, as well as impedance matching and buffer circuitry. Also, the patching system is capable of mixing and combining signals, which is difficult with a conventional patch bay.

Lost control settings can be a real problem. You might spend an entire day getting a really beautiful mix. The client then leaves with this mix, only to find while listening to it at home that he forgot to boost something. If you haven't written down all the control settings, you have no choice but to re-mix from scratch. Consoles need a memory to help us get those controls back in the same place. I mentioned earlier that controls should light up when in use; even better, they should *refuse* to light up until repositioned to settings held in a memory. This system can be bypassed when desired.

The cold, sterile, intimidating appearance of audio consoles can negatively affect the atmosphere within the studio. Unfortunately, most manufacturers use a profusion of plastic knobs and switches, and painted metal with silkscreened lettering. More imaginative use of materials is needed. Console designers need to realize that we are in a glamour industry and esoterica is very important.

Another real problem that we encounter is the difficulty in matching professional audio consoles to semi-pro or non-standard gear. I suggest the inclusion of a small plug-in circuit board with the appropriate programmable resistor-IC positions to compensate for differences in equipment. Once these PC boards are programmed, any piece of equipment can be patched to any other piece of equipment in the system without concern for equipment differences.

The last problem I wish to discuss is overly-glorified specifications. Exaggerated specifications can fool us into buying something we wouldn't buy if we knew the truth. The solution to this problem is for manufacturers to take their equipment to independent laboratories for testing and agree to publish these results along with the name and address of the person conducting the test so that interested parties may inquire into the details. Also, more complete information about the test conditions needs to be published with the specifications.

The diagram on the following page shows how the ICC console approaches some of the problems and solutions mentioned above. ■

Introducing
the audio processor that remembers.

Super Prime Time™

The first **Microprocessor-based** special effects device.
It has non-volatile memory storage for special effects . . .
recalls them on command . . . in the studio or **on stage.**
Now, with Lexicon's Super Prime Time digital delay — **the first programmable, microprocessor-based audio processor — you can create . . . store . . . and recall the desired effects for any given piece of music, as well as have a** frequency response of **20 Hz to 20kHz** at all delay settings.
There's never been an audio processor like it.
For the performer, Super Prime Time opens a whole new world of virtually unlimited musical enhancements. This remarkable system stores 8 factory preset programs and 32 user programs of effects . . . recallable at any time.
Programs may be off-loaded to audio tapes or cassettes providing unlimited off-line storage.
Super Prime Time. The first microprocessor-based delay . . . the audio processor that remembers.
Write for full details today.

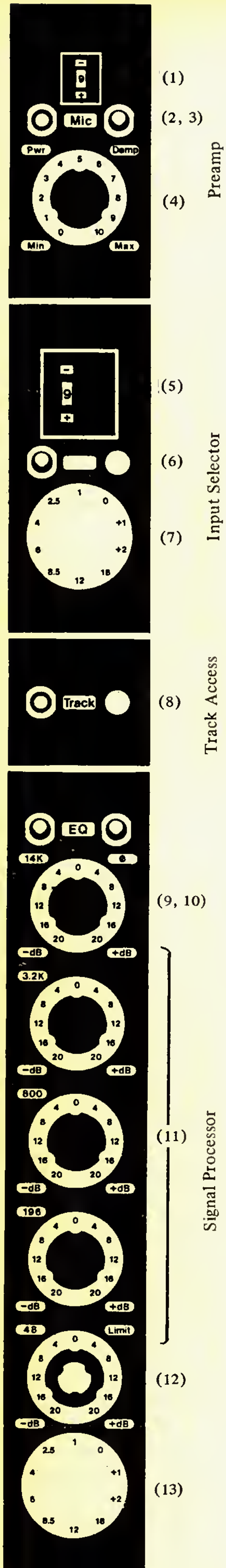
Lexicon

Lexicon, Inc., 60 Turner Street, Waltham, MA 02154/(617) 891-6790/TELEX 923468
Export: Gotham Export Corporation, New York, NY 10014

Space photo courtesy of NASA

Circle 19 on Reader Service Card

I.C.C. CONSOLE: CHANNEL



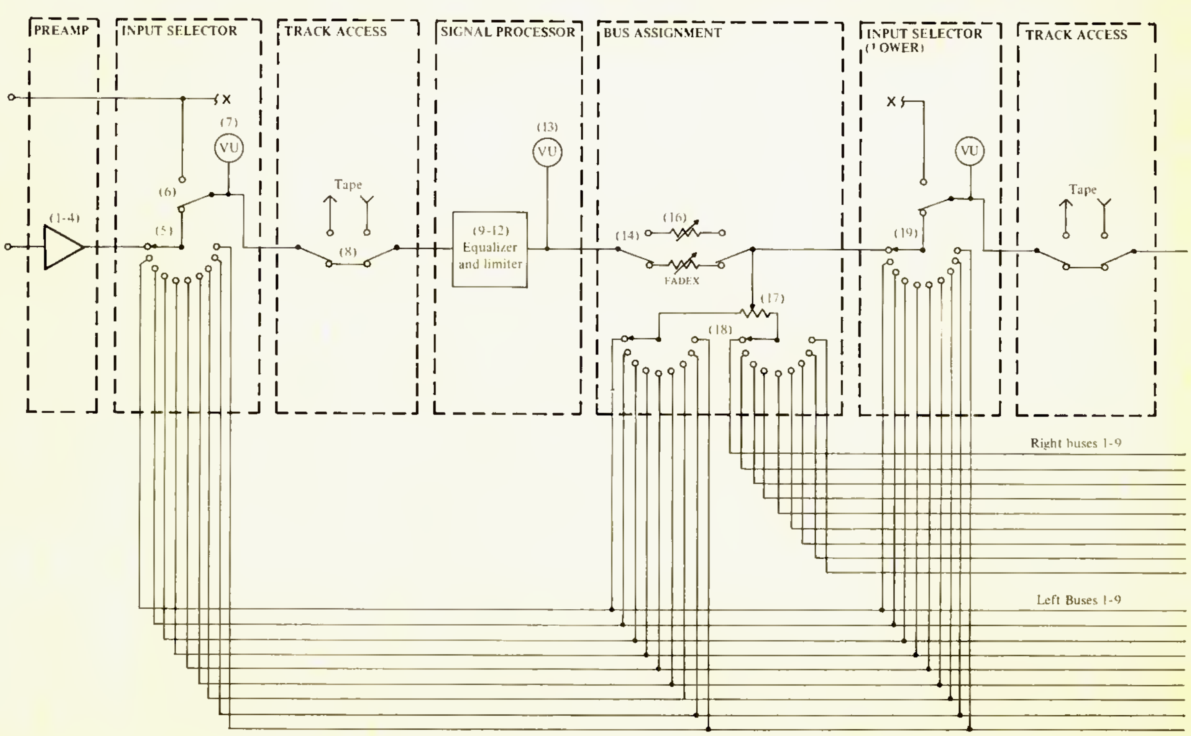
PREAMP. This module contains peak compressor (1), preamp activate switch/ phantom power (2), compressor switch (3), and preamp gain control (4).

INPUT SELECTOR. Odd-numbered channels have a Left-Bus selector switch, and even-numbered channels have a Right-Bus selector switch (5). Buses are free-floating, and may be used as desired. When the switch is in position 0, no bus is selected, and the preamp modules output is routed to the input selector. The Line Selector switch (6) over-rides the selected bus/preamp input. The output of the module is monitored by a 10-segment LED meter (7).

TRACK ACCESS. When the Track Access switch (8) is up, the tape recorded is inserted at this point in the signal path.

SIGNAL PROCESSOR. This module contains a 10:1 limiter, phase reversal switch, equalizer and VU meter (9-13).

*Channel Block Diagram. This block diagram has been prepared by the editorial staff of **db** to illustrate the signal*



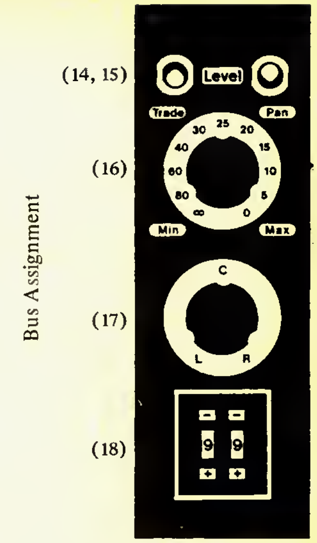
MODULES AND BLOCK DIAGRAMS

BUS ASSIGNMENT. A two-position switch (14) replaces the Level Control (16) with the VCA (fadex) fader. A second switch (15) routes the signal directly, or through a pan pot (17) to the bus(es) selected by the Bus Selector switch (18).

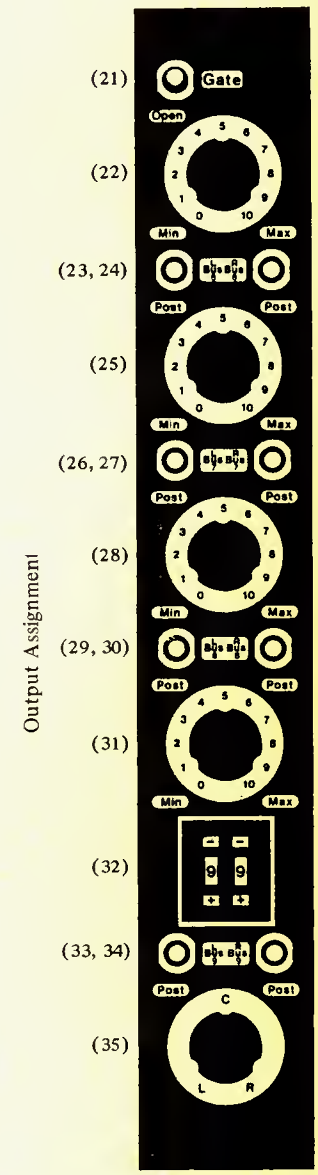
LOWER INPUT SELECTOR and **TRACK ACCESS** (shown in block diagram only). These modules performs the same functions as the Input Selector and Track Access modules described above.

OUTPUT ASSIGNMENT. A two-position switch (21) inserts a noise gate whose threshold whose threshold may be varied by potentiometer (23). A series of switches (23, 24, 26, 27, 29, 30) and potentiometers (22, 25, 28, 31) route the signal to Left and/or Right Buses 6, 7, and 8. Alternatively, the Bus selector Switch (32) may be used to route the signal to any Left and/or Right bus. This signal may be sent directly to the buses, or through the Bus 8 Send controls (29-31). A pan pot (35) allows Left and Right Buses 9 to be used for the final stereo mixdown.

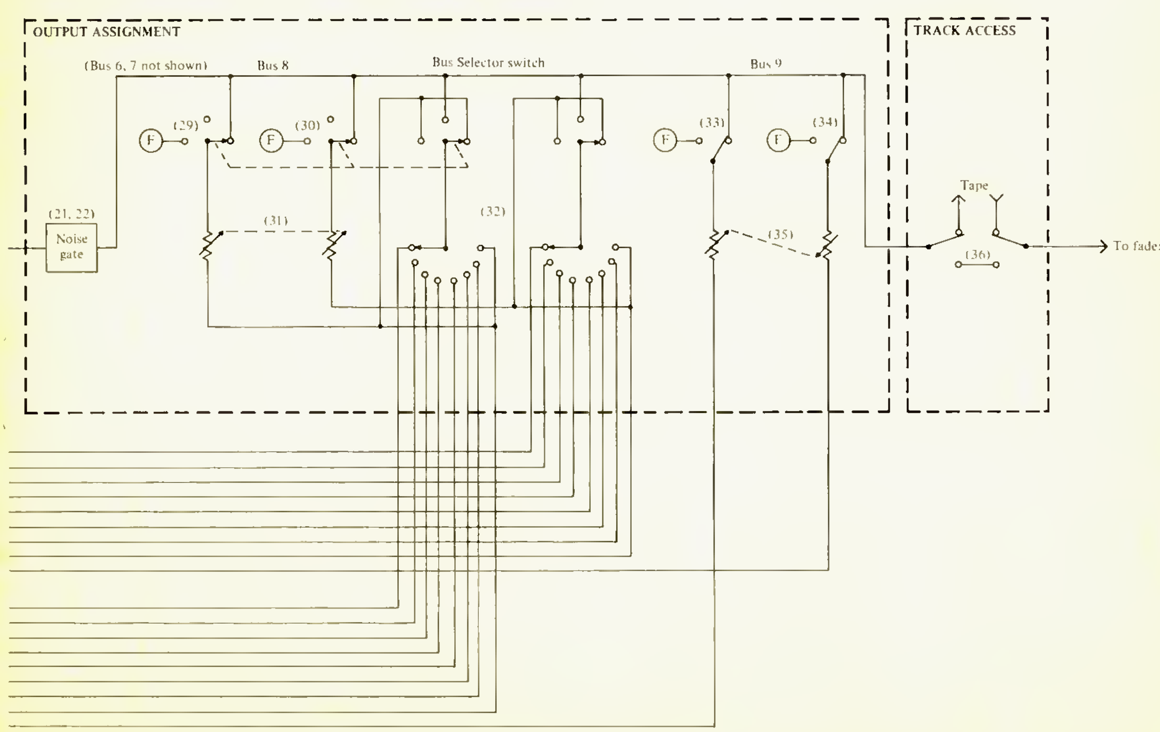
TRACK ACCESS. This module duplicates the previous Track Access modules, and allows the tape recorder to be inserted at this point in the signal path. The output of this module is routed to the fader and then to the monitor system. However, if the Trade Switch (14) on the Bus Assignment module has been activated, the fader is re-assigned to the Bus Assignment module, and is replaced by the Level Control potentiometer (16) from this module.



Lower Input Selector and Track Access not shown



flow through the channel modules. It is not necessarily a faithful representation of the actual switching system.



FOR THE BRIGHTEST STARS.

The talents of recording stars and their studios provide the magic that turns a reel of recording tape into an outstanding creative achievement.

The Ampex Golden Reel Award honors those achievements which were mastered on Ampex professional recording tape. They have earned a place in the ranks of the world's most successful recorded albums and singles.*

Included in the award is a gift of $1,000 to the recording artist's favorite charity.

Over the past three years, there have been more than 200 Golden Reel recipients. And more than $200,000 donated on their behalf.

Congratulations to all of them on a masterful performance.

AMPEX

Ampex Corporation, Magnetic Tape Division, 401 Broadway, Redwood City, California 94063 415/367-3889

 Golden Reel Winners as of 1/81

Circle 35 on Reader Service Card

*RIAA Certified Gold.

IRVING KAUFMAN

Inside National Video Center & Recording Studios: 42nd Street's Newest Star

Here, we take an inside look at one of New York's most successful and up-to-date audio/video recording studios.

National Video Center & Recording Studios partner/co-founder Irving Kaufman has as much to look forward to as he does to remember. An audio engineer for close to 40 years, Kaufman's first professional association was with WNEW AM Radio soon after his graduation from Rutgers with a degree in physics. Branching out as a consultant and designer for many of NYC's early recording studios, Kaufman joined Nola Sound in 1945, where he engineered hundreds of lp and single sessions for such stars as Perry Como, Louis Armstrong, Benny Goodman, etc.

In 1952, he became chief engineer for NY's Audio-Video Studios, where he specialized in commercial recording sessions. (At one point, more than half the radio commercials aired in NYC were engineered by Kaufman.) When A-V Studios closed in 1959, he and partner Hal Lustig pooled their resources to take over their former employer's facilities at 730 Fifth Ave. and opened National Recording Studios.

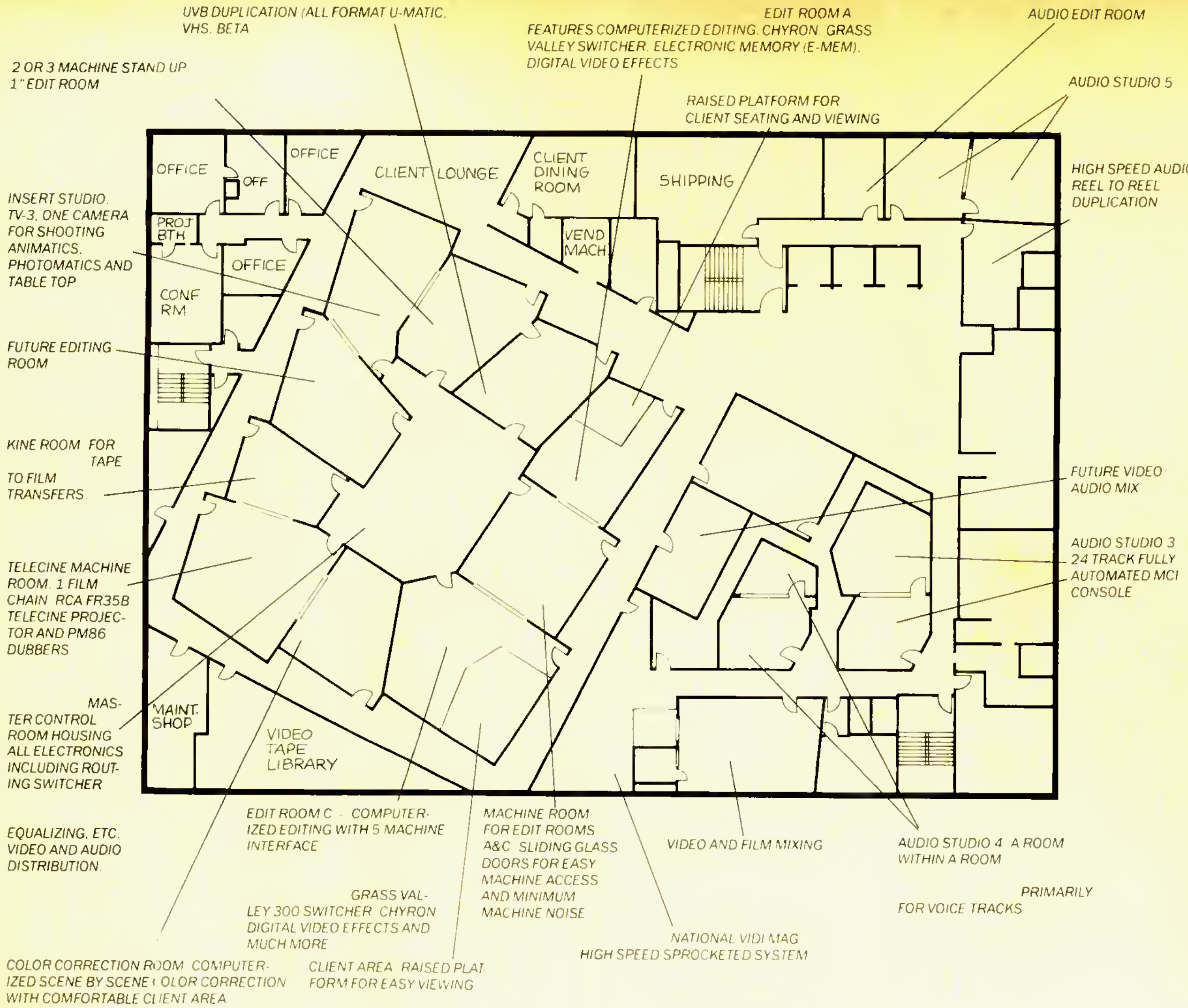
THE ORIGINAL NATIONAL RECORDING STUDIOS consisted of three studios in a cramped office building. From our earliest days, we established ourselves as a fast, reliable facility. This reputation was responsible for considerable repeat business and for a steady stream of word-of-mouth work.

Our growing work load necessitated expansion in two areas: whenever additional physical space became available at 730 Fifth, we were quick to take advantage of it; we also required skilled, dedicated technicians, and we developed a program of training and promoting from within our ranks.

Many of National's technicians have been with us 10, 15 years or more and, as part of our "family," they are largely responsible for our success. For example, senior mixer Dick Mack introduced the film mixing division to National. Mac Anderson, our first messenger, is now a key audio engineer. Rod Zavala also began as a messenger here 18 years ago, and has helped the film division become one of our most successful operations. Chief audio engineer Eldo Luciani, a 19-year veteran, takes much of the credit for the selection and reliability of our equipment, and the list goes on.

Keeping abreast with the times and with our clients' changing needs prompted National to expand our audio recording facility to include film mixing some 20 years ago. In 1973, as videotape production and editing became the obvious wave of the future, National geared up to service clients in this area.

Irving Kaufman is the co-founder of National Video Center & Recording Studios.



By 1978, we had completely outgrown our 730 Fifth Ave. facility and we began looking in earnest for a new home. After much searching, we learned that Manhattan's West Side Airline Terminal Building was available. Chief video engineer Herb Ohlandt, our staff and I spent the next two years working on the design for the new National and, in April 1981, we finally made our move.

Located at 460 West 42nd Street, in the heart of one of New York's most ambitious neighborhood revitalization projects, our one-stop studios feature state-of-the-art audio and video technology. This area teams with growth and optimism: the New York Convention Center (now under construction); Manhattan Plaza, a high-rise multi-building apartment complex; a row of modern Off-Off Broadway theaters; and several very good restaurants have resulted in people comparing the resurgence of West 42nd Street to the kind of rejuvenation that took place uptown with the construction of Lincoln Center more than 20 years ago.

An enormous structure occupying almost half a city block, the West Side Airline Terminal provides National with nearly five times the studio, production, editing and client conference space we had at our Fifth Avenue location. We've grown from about 12,000 square feet to more than 55,000 square feet, spread over three floors.

THE FIRST FLOOR

Originally, the Airline Terminal utilized the wide-open ground floor expanse as a passenger gathering space, ringed by ticket counters. It was a logical decision to convert this vast space into our two shooting stages.

Approximately 10,000 square feet have been developed into two large video studios (TV-1 with 4,000 square feet; and TV-2 with 2,000 square feet); reception and lobby areas; separate make-up and dressing rooms for each studio; a green room; conference room; production office; and a scenery/paint shop. Also on our first floor are storage areas and special access ways to the street permitting direct deliveries of scenery and props. There is a drive-in access to TV-1, and a convenient off-street adjacent parking lot.

Designed for original video productions, the two shooting stages have their own individual control rooms. Additionally, TV-1 offers two-cornered hard cyc capability, a separate audio control room, and a computer-controlled lighting grid with individual dimmers. Both spaces are ideal for either feature film or TV productions, and TV commercial shoots necessitating a large stage.

National Video Center Edit Room "C" featuring Computerized Editing with 5 machine interface, CMX 340, Grass Valley 300 Switcher and Digital Video Effects & Chyron Character Generator (not shown). Edit room "C" is one of the largest editing rooms in the U.S., and is designed for long show-editing sessions.

National Video Center's Edit Room "A" featuring computerized editing, Chryon, Grass Valley 1600 Switcher equipped with Electronic Memory (E-MEM), Digital Video Effects (DVE) Grass Valley Mark II.

THE MEZZANINE

Our second floor mezzanine provides National with corporate executive offices and suites reserved for incoming production companies. Few, if any, New York facilities offer the luxury of client office space; and with an additional 5,000 square feet available on this floor for future expansion, we're considering all eventualities so that we'll have the flexibility to move in new directions later on.

For example, we may move into digital audio recording, though we're biding our time until the digital scene settles down and becomes standardized. We credit much of our long-term success to this somewhat cautious approach to new technologies.

THE THIRD FLOOR

Our third floor design should give visitors a sense of the efficient flow of audio/video post-production activity at National. A bank of elevators opens into a huge reception area backed by a common scheduling room for both audio and video services. Adjacent to the scheduling room is a series of audio studios which Eldo Luciani, our vp in charge of audio engineering, and I designed. These include a fully automated MCI-equipped music studio; two states-of-the-art acoustically designed studios primarily for voice tracks; and an elaborate video and film mixing theater supervised by our senior mixer, Dick Mack.

All our audio recording studios are outfitted with UREI time-aligned speakers, which our engineers agree provide a much cleaner sound than comparable units, especially when the sound level is low. The studios are also equipped with a large variety of popular microphones.

Some facilities will always champion their "favorite" manufacturer because of a string of "good luck" experiences. Realistically, however, I feel that with the variety of fine equipment available these days, the performance of a specific piece of hardware is only as good as the engineer who employs it. Our training concentration is on engineering, not equipment.

VIDI MAG

Dick Mack's pride and joy is our unique Vidi Mag System (which was developed by Bob Fine of Magna Tech, with our assistance in the electronic area). Vidi Mag is a sprocket-driven videotape recorder that facilitates high-speed audio editing and mixing. Similar to the double system handling techniques used for over 50 years in motion picture post-production, Vidi Mag allows greater flexibility and creative control in mixing audio against picture.

Because Vidi Mag interlocks with dubbers, the picture always remains in precise sync with the sound tracks. And since no pre-roll is required, the system offers a distinct advantage for recording and mixing several tracks at a much faster pace. Vidi Mag may be interfaced with any modern editorial set-up, permitting film editors to lay out sound tracks in their cutting rooms—not at a mix session.

Though other post-production houses have access to Vidi Mag, National is New York's only sound studio with a full companion video facility. As a result, Dick Mack has been busy with work for major advertising agencies like BBDO, J. Walter Thompson, Kenyon & Eckhardt, Benton & Bowles and Ogilvy & Mather, and nationally known products including Pepsi Cola, Chrysler and General Electric.

Also located in this studio area is our high-speed audio reel-to-reel duplication room, two audio edit rooms, and several rooms, as yet unoccupied, one of which most likely will become a second audio/video mixing facility.

New York's West Side Airlines Terminal building prior to its metamorphosis into National Video Center & Recording Studios Inc. This scene, shot from the 10th Avenue perspective, is currently National Video's TV 1, a 4,000 sq. ft. video production studio.

THE VIDEO TRIADS

To the right of the reception area and scheduling room is our video post-production area. Arranged in *three triads*, all nine rooms rings *a central master control room* housing complete electronics including a routing switcher which eliminates patch panels, and the various electrical juggling acts that accompany them. Our video post-production complex is the brainchild of Herb Ohlandt, vp of video engineering.

One triad houses Edit Room A featuring a Datatron Vanguard one-inch computerized editing system complete with Chryon, Grass Valley Switcher, Electronic Memory (E-Mem) and Grass Valley Mark II Digital Video Effects (DVE) Generator. It was with the DVE that National recently created a number of striking visual effects—some never before seen on TV—for a one-hour cable stereo special featuring MCA recording group Spryro-Gyra, a popular jazz fusion sextet. The spectacular, animated video paintings prompted a *Variety* reporter to predict a revolution in future video effects projects for music groups.

Across from Edit Room A, Edit Room C provides computerized CMX editing with five-machine interface. Both one-inch and two-inch tape is edited in this comprehensive set-up which features digital video effects and the much-discussed Grass Valley 300 Switcher. Both edit rooms have raised platforms for comfortable client viewing, and are linked by a central machine room with sliding glass doors for easy machine access and minimum machine noise.

The second triad is made up of a color correction room offering computerized scene-by-scene color correction; a telecine machine room equipped with RCA FR35B projector and PM86 dubbers; and a kine room for fast, clean tape-to-film transfers.

The third triad includes a UVB duplication room with the capability of handling all formats (U-Matic, VHS and Beta); and insert studio designed for animatics shooting and tabletop work; and space reserved for a future editing room. (A recent segment of CBS-TV's *60 Minutes*, devoted to the complex and extensive research involved in test-marketing Stroh's Beer, featured two 30-second animatics produced by Creative Ways (NY) that were completed in National's animatics room.)

Also included in our third floor are executive offices, shipping rooms, a tape library, client lounge and dining room, conference room and two maintenance shops.

ANCILLARY FACILITIES

Although National has invested close to $7 million in developing our West 42nd Street complex, we have maintained our facilities at The National Film Center on Manhattan's East Side; the high-speed cassette duplication operation in Long Island City; and our fully equipped Edison Hall music recording studio in the Edison Hotel on West 47th Street. With the demise of Columbia Records' 30th Street Studios, the Edison Hall studio is one of a handful of facilities in New York large enough to accommodate a full-size concert orchestra.

CBS CABLE AFFILIATION

Several of our recent projects have been especially exciting. Showtime, ABC Video Enterprises, Warner Communications and the Bravo Cable Network have all utilized National for a variety of production and post-production work, which have included the participation of such notable performers as Ed Asner, Orson Bean and Rita Moreno.

But, undoubtedly, our most prestigious coup to date was being selected by CBS Cable as the production and post-production facility for its initial series of cultural programs. Hosted by award-winning broadcaster Patrick Wilson, the shows include lavish productions and interviews with prominent figures from the worlds of theater, cinema, dance, the fine arts and science. As we enter the age of cable TV, we expect to be handling a good deal more of this kind of work down the road.

The audio and video recording industries have continued to grow enormously, opening up a wealth of opportunities for artists, engineers and production and post-production facilities. I'm confident that the new National is ready for the challenge. ■

ALAN FIERSTEIN

The Anatomy of a Custom Console: The Acoustilog GB-1

The GB-1 is yet another example of the old adage, "If you want something done right, do it yourself."

ALTHOUGH THERE ARE MANY reasons why custom-built consoles show up in studios, the deciding factor is usually the price/performance ratio. Mass-manufactured consoles have the edge on features per console dollar: custom console performance can be optimized for one studio and its style of recording. If the studio wants the cleanest, least-processed sound, it can, of course, order a stripped-down no-frills board and probably save money while increasing performance. Many studios customize their boards to an extent, but this is much easier when the special controls they desire can be located in the most accessible spot on the board. Of course, some special features can only be built into a non-stock frame right at the outset of construction. All of these factors led to Acoustilog building the GB-1 console—shown in FIGURE 1 for its own studio, Sorcerer Sound.

Sorcerer Sound is, in addition to a recording studio, a demonstration and research room for Acoustilog. We develop new recording equipment to our own needs and later find that our customers have been looking for the same type of device. Since we manufacture custom consoles, it was only natural for us to install a console with many unique features that our customers could choose from.

DESIGN CONCEPTS

For signal purity, there is very little circuitry in the console. Only two necessary active stages per channel are necessary: the mike preamp and the VCA. Of course, the VCA can be patched around during recording. The VCA used is the Valley People EGC-101, in a Fadex-like automation configuration. It was originally planned to install individual channel VCA bypass switches; however, the prototype module was so clean and quiet that we decided the VCAs could remain in the circuit permanently. (For more details on the EGC-101, see "The New VCA Technology" in the August, 1980 **db** — Ed.)

FETs allow very flexible signal routing but also add to the number of active elements, each with its own small-but-significant distortion. Therefore, no FETs were used, except in the monitoring system. Relays provide excellent signal handling when clean but can become a reliability problem after a few months in the city, so none were used. There are no trimpots in the signal path. These can also become noisy or distort, especially when not adjusted over a long period of time. Precision resistors eliminate the need for trimmers.

Heat is one of the greatest enemies of electronic equipment. It deteriorates semiconductors and capacitors, hastening their failure. We avoided excessive operating temperature in several ways. First, we use minimal circuitry, and low-current amplifier chips. Second, as mentioned before, no relays were used. The incandescent lights in the console are run on low voltage, and LEDs are used sparingly. Most indication is on "shadow" type switches, which use a mechanical window to display a fluorescent dot when the switch is depressed. The result of these efforts is a console which runs so cool, it is not possible to feel heat through the top panels. The console requires less current to run, and therefore less air conditioning to cool, providing a double savings in electricity.

Alan Fierstein is the president of Acoustilog, Inc.

Figure 1. Two views of the GB-1 console at Al Fierstein's Sorcerer Sound Studios.

As we do maintenance at many studios, we get to see what types of connectors cause the most problems. We have witnessed the widespread use, and failure, of a particular type of connector consisting of a bent tin spring and mating pin in a cheap plastic housing. The GB-1 console uses only bifurcated, gold-plated edge connectors for the audio paths.

The console itself can be a source of serious acoustical problems in control rooms. Many modern-generation automated consoles dip so deeply down to the floor that, from the elevation standpoint, the control room is divided into a front and a back half. This always weakens the bass response behind the console. Proper studio design can minimize, but not eliminate, the severity of this effect. An isometric view of the GB-1 is shown in FIGURE 2. There is enough room under the console to allow you to crawl around on your hands and knees, rather than on your belly like a slithering reptile. Thus, maintenance is simplified, and acoustical problems are almost nil.

Speaking of acoustical problems, it has been the author's experience that many control room acoustical ills can be traced to resonating panels in consoles. Dampening out these ringing gongs requires liberal use of visco-elastic damping compound, but why should this be done by the console buyer? Simply by using thicker metal and wood in the construction, audible resonances are stopped. Tape machine manufacturers take note!

Perhaps the most striking chief feature of the GB-1 is the shape. The V-wing design, by Sorcerer engineer Greg Curry, is more expensive to implement, but it provides several distinct benefits. First, the console shape naturally guides the engineer to the room center. Those of you who have to remember that the room center is "a quarter-of-an-inch to the right of the edge of the fader knob on channel 20" can appreciate this. The many vertical compartments hold auxiliary equipment, meters, patch bay and future ideas at an easy-to-reach position. The separate engineer's and producer's desks offer ample space for food and equipment. Accidental spills are usually not a problem, because the desks are located below the level of the electronics, and the entire structure is covered with acid-resistant formica, originally designed for chemical laboratories. The leather armrest is replaceable in three sections, should a cigarette land in the wrong place.

OPERATIONAL FUNCTIONS

The features of the modules are fairly standard. The usual brigade of sends, switching, and equalization are there, in addition to a peak detector which senses clipping anywhere in the module, and at three solo points and solo-in-place. "Fader on" indicators show levels greater than −85 dB. Channel On-Off Switch/lights indicate which channel is on.

The On-Off Switch is of the soft-switching type, completely free of clicks. The light in the switch is off when the Switch is off (which is the un-pressed position), or when the VCA in the fader is below −80, which could be for several reasons, such as its pot being down, its master being down, or another fader being soloed. If the Master of a group is turned off or faded out, it will of course turn off or fade out the slaves in that group, and their lights will then go out. These are normal features of the Fadex system, by the way.

The two previewing solo systems are controlled by the Master Solo button. When it is pushed in, solo is on, and pushing a solo button will immediately activate the solo system. When it is not pushed in, you are in the "solo ready" mode. You can preset the solo buttons you want and then solo the entire set by pressing the Master Solo button. If you are in Solo, the Solo button and the Channel number LED above the soloed channel or channels will light up steadily. If you are in solo ready, and a solo button is pushed, the Solo button and the Channel number LED(s) will flash. If the Solo system is not on, all the Channel number LEDs are lit, easily identifying the channels.

A few unusual features in the communication system makes life easier in the control room. For example, Cue-Talkback is a locking button that allows you to talk to the musicians cue system, without muting the monitors as the normal talkback does. The locking feature is especially nice when the producer wants constant communication with the artist. ICM (Monitor Intercom), a function we have incorporated into our boards since 1974, comprises a stereo intercom system that is, in essence, a pair of always-ready room mikes. Solo ICM is the same thing, but allows 2-way conversation when talkback is in effect. Feedback is possible in this mode, although a special feature of the GB-1 is an automatic monitor protector called the POP-110.

The POP-110 system is preset for monitor speaker maximum level, and instantaneous peak sound levels which exceed the threshold are heavily attenuated for a period of one second (longer if the excessive level is continued). The speaker protection system is in the signal path only during the protection mode, and there is no active circuitry in the signal path during normal operation. Peak detection is used for extremely fast response time. Many mixers are surprised to learn that large

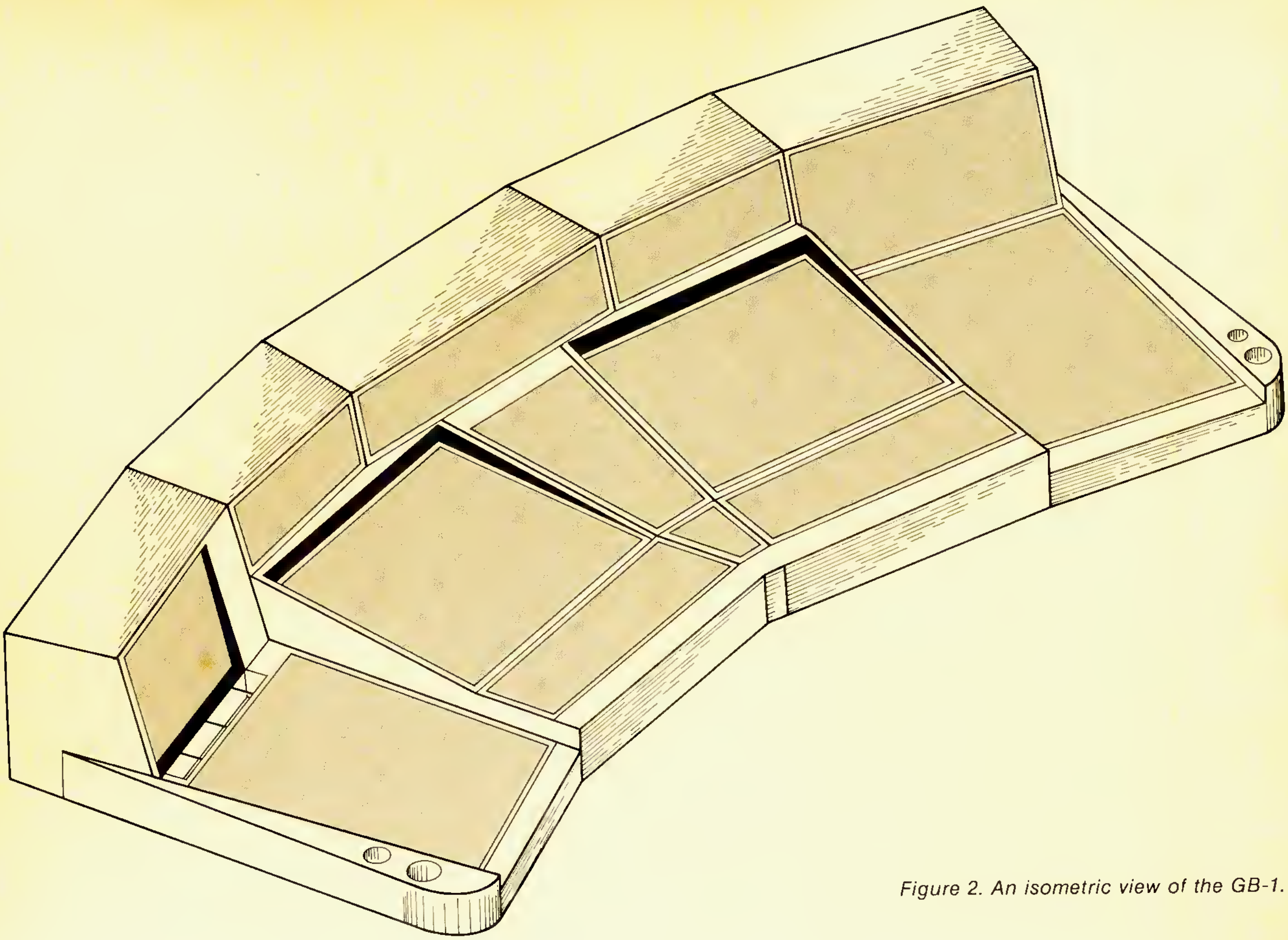


Figure 2. An isometric view of the GB-1.

blasts of sound can permanently change the response of the monitors, even if speaker damage (or ear damage) doesn't occur. These transients can occur, for example, when a tape channel in the input mode feeds back through a line input of the console. Sound familiar? (The POP-110 has proven so successful that we are releasing it as a separate new product.)

A compact set of buttons next to the Control Room Monitor level control allow listening in mono to either or both of the two channels as well as muting of the two monitors. Three monitor systems are selectable, and a Phase reverse button (as it is commonly known) flips the polarity of one monitor channel. When you press this button, mono information should lose bass and imaging should spread out to the sides. If not, your speakers are wired "out-of-phase." Naturally, this switch is a momentary button.

MULTILYZER

The Multilyzer, designed by Ted Rothstein, is a combination multichannel meter system and 1/3 octave spectrum analyzer. It is usually a meter, switching to Analyzer when the Solo systems are active. When this occurs, you see the spectrum of what you're soloing, as well as hearing it over the speakers. This helps you to properly equalize sounds. The Monitor Spectrum switch (on the Master Panel) also activates the Analyzer mode, giving an analysis of what you're listening to on the mix of all channels.

As a metering system, you have a choice of Average, Peak, or Both simultaneously. Slow and Fast modes are available, as are three scales: 60 dB log, 20 dB log and VU. You can mix and match modes and scales to the Average or Peak or Both. The Both mode is very useful during most operations. If the same scale (such as 60 dB log) is selected for both Average and Peak sections, you will constantly see the peak/average ratio over a tremendous dynamic range. Watching the Average will allow you to identify tracks and see when they come in and out, and watching the Peak mode (which has a slow decay feature) will allow you to record the highest possible clean signal on tape, by not allowing the peaks to go more than 13 dB into the red (above 0 VU). This takes into account the 13 dB headroom above 261 nWb/M flux level that is typical on tape. When observing the Multilyzer display, we are constantly amazed at how high you can record non-percussive tracks, and at how low you must record transient sounds like triangle and cowbell. This cannot be done accurately with normal VU meters.

The meters are connected to the line inputs of the console. Since the tape machine normals into the line inputs, you will always have Multilyzer activity, whether you're recording (tape machine on input or sync) or mixing (tape machine on playback).

A WORD ABOUT QUALITY

The console reflects our view that "less is best." Very few active stages are employed in each channel. While this eliminates some features many of us are used to, it has several advantages, like ease of operation, cool-running operation, higher reliability and improved signal purity.

P.S. A special word of thanks to Paul Buff and Tom Behrens for helping in the design of the fader system. ■

LINDA JACOBSON

Electronic Synchronization in the Recording Studio

When it comes to time code synchronization, the Shadow knows…

ATTENTION ALL RECORDING engineers: Away with those razor blades! Electronic editing has successfully infiltrated the sound industry, and one of the most potent techniques is digital synchronization (also known as time code synchronization).

To synchronize is to mechanically establish an active, running relationship between the various multi-track tape machines used in a production. Synchronizing one tape machine to another facilitates precise editing, mix-down and dubbing. Audio work frequently involves varying the phasing between tapes, a process made simple by digital synchronization. Synchronizing your multi-track audio tape recorders dramatically increases the number of available tracks.

In a typical recording studio appliction of the synchronizer, *all* original session material may be recorded on a *single* multi-track machine. When the engineer runs out of tracks, a rough mono mix may be dubbed to a two-track machine, which is used as a synchronized reference playback during the recording of additional tracks (on the same multi-track machine). The result: 30 or more audio tracks laid down sequentially using only one recorder (plus one track on each tape to provide time-code indexing). At final mix-down, the two masters would play to a single board from two synchronized transports (which might be in different control rooms). The dramatic increase in channel capacity is attained at the one-time cost of a synchronizer.

The video industry was responsible for the introduction of the electronic synchronizer. The use of this technology within the audio industry can only benefit the latter: With today's increasing desire to improve the quality of television audio, the sound studio can increase its market by acquiring the tools and knowledge necessary for the audio/video interface.

In addition, being able to synchronize ATRs to VTRs allows the studio to offer audio/video production facilities to television producers, and simultaneously increases multi-track studio sound recording capability. Also, many studios are using video as a marketing tool for recording artists; more and more musicians are using video demos to attract a label's attention and create an image.

The newest synchronizers easily provide a precise lock between high quality audio and standard video programming. One such interface involves a four-track, half-inch digital audio tape recorder locked to a one-inch type-C VTR. This application is ideal for stereo simulcasts, in which optimum audio quality is a must. (Actually, it's good for all types of standard TV programming.)

Electronic synchronizers are microprocessor-based devices which control the capstan motors of tape machines. The most popular ones (manufactured by such companies as BTX, Audio-Kinetics, MCI, Adams-Smith, and EECO) are synchronizers that perform by reading SMPTE/EBU time code.

Linda Jacobson is the media coordinator for the BTX Corporation, Natick, Massachusetts.

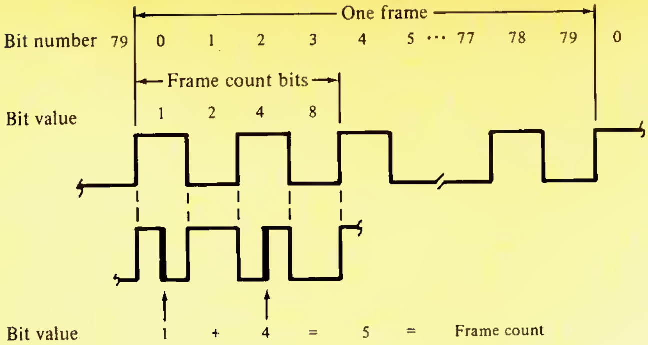


Figure 1. Detail drawing of the SMPTE time-code system. Note the level-switching during the middle of bits 0 and 2 (indicated by a heavy line width in the drawing). The corresponding bit values are added (1 +4) to determine the frame count. Other bit numbers indicate hours, minutes and seconds.

TIME CODE

SMPTE time code was introduced specifically for video editing in the mid-70s by the Society of Motion Picture and Television Engineers. It's a binary electronic signal (i.e., a square wave) recorded on a separate track. The code precisely identifies each 1/30-second segment of tape with 80 bits of information, as seen in FIGURE 1. The 80 bits are numbered from 0 to 79, and are divided into groups that define each segment, or frame, with eight digits representing hours, minutes, seconds and frames. Timing is on an absolute 24-hour system, with 30 frames per real second (USA standard; the European EBU standard is 25 frames per second). Essentially, time code is the digital equivalent of the physical sprocket holes in film. The time code data is always directly related to the tape position, and therefore it can provide the very precise indexing needed for sophisticated editing. (For more details, see "SMPTE Time Code Comes To Audio" in the November, 1978 **db**—Ed.)

Using a time code generator/reader in conjunction with a synchronizer is ideal; this way the code from an original master tape can be read and identically regenerated onto a work tape while the two are running on separate machines. The result is two encoded tapes that can be edited to accuracies within a fraction of a frame (typically less than 1/30 second) on electronically-synched machines.

A synchronizer provides the ultimate in production flexibility, since it takes the engineer's hands off the tape machines and allows him to concentrate on creating top-rate material. An electronic synchronizer initially controls the capstan servo of the slave machine by time code address comparisons, driving the capstan by signals proportional to phasing errors until frame sync is achieved. The synchronizer controls the capstan until the slave's phase error is adjusted to less than 1/100th of a TV frame, thereby locking the slave to the master.

While "simple" electronic synchronizers function only in the record/playback mode, more complex systems can provide full control of all tape motion from a single keyboard. Into the second category falls the new BTX Shadow System.

THE SHADOW SYSTEM

Introduced this past April by The BTX Corporation, the Shadow is an "intelligent" SMPTE/EBU time code synchronizer and tape machine controller. It interfaces not only audio and video tape recorders, but sprocketed film dubbers, computerized mixing consoles, and video editors.

An important feature of the Shadow is a standardized RS.232C communications channel, which provides the studio with the ability to execute complex editing and control functions from a remote computer. Virtually any commercial micro-or minicomputer can be used to control and synchronize both master and slave transports via the Shadow, offering control capabilities of great accuracy and flexibility, not to mention sophistication.

The computer interface enables instant logging of cues and specific tape locations. For example, time code data, user bits data, current status information, and learned data (including indication of preselected, flagged points) may be transmitted at a 30 Hz rate. Most important, the RS.232C interface provides a link between low-cost, highly functional computer technology and the increasing needs of audio production studios.

With RS.232C implementation, the Shadow synchronizer provides full "chasing" capability, while also being able to synchronize precisely to the subframe (1/100th frame). (Most video editing systems deal only in units as small as frames, each lasting a 30th of a second, which is simply not good enough for complex audio editing.) When the machines are in Fast Forward or Rewind, the Chase feature enables the slave to shadow its master; this minimizes lock-up time when in Play, since a minimum differential between the two machines was maintained during the fast wind mode.

The Shadow's most unusual characteristic is a fully-adaptive controlling scheme that uses a special-control feedback loop. This enables the Shadow to "learn" as it servos the slave machine, trimming its commands to match the dynamics of the transport. The result is a very fast lock-up time with minimum overshoot, and automatic compensation for variations in hold-back tension and tape mass.

The Shadow's adaptive controlling scheme constantly measures the characteristics of the slave machine (its rate of frequency response to a control signal), which determines the characteristics and behavior of the servo control loop. The Shadow adapts the control loop to best suit the situation, which allows optimum performance from each tape machine. Synchronization performance is limited only by the constraints of capstan servo response and internal transport noise sources.

The Shadow accomplishes synchronization by over-riding the transport capstan control, with either variable voltage or frequency. The range of that over-ride signal depends upon the dynamic characteristics of the particular machine connected to the Shadow. So, the Shadow must be told what voltage or frequency is needed to slow down or speed up the slave machine in order to lock it to the master.

The control loop response is optimized for the specific transport mechanism by programming initial slave machine characteristics into the Shadow before operation.

This process is a relatively simple one: It involves the adjustment of two potentiometers located inside the Shadow (as per the instructions given in the Shadow Operator's Manual), in order to set the high and low limits of the servo. The top cover of the Shadow lifts off easily to facilitate calibration. As a result of the adaptive loop, the Shadow learns—and remembers—the dynamic characteristics of the transport mechanism, for the best possible response time in all applications.

The Shadow's digital-to-analog program is controlled via use of two semi-cascaded, 12-bit D/A converters which control the slave machine's speed accurately to 0.0003 percent of nominal play speed. Such control is vital at times when the Shadow's feedback loop is broken (by tape drop-out or unreadable time code), and control must be kept to maintain precise synchronization.

A

B

Figure 2. The BTX Shadow System Synchronizer for video and audio tape recorders (A), and the System Command Console (B).

The combination of the Shadow's adaptive control loop and digital-to-analog converters result in synchronization precise to one three-thousandth of a second (that's within 0.01 inches of tape moving at 30 i.p.s.). This type of fine resolution is ideal for synchronizing audio to video where precise, between-the-frame audio cuts are required.

The basic Shadow system is a one-box chasing synchronizer with only two controls. However, the system may be upgraded to a tape machine controller/auto locator, by using an optional Command Console, which has 34 dedicated keys and a 10-digit LED display for the receipt and transmission of data. The console is shown in FIGURE 2.

The Shadow with the console (*or* when interfaced via RS.232C) gives the sound engineer remote control over the major tape machine functions. The console's display shows master or slave time code (regular or drop frame), as well as slave offset data. When operating the Shadow as a controller, both master and slave transports can be commanded to locate ("GO TO") a specific location entered via the keys, or a location already stored in one of the Shadow's nine "scratchpad" memory registers. In addition, the engineer can create and maintain up to 24-hour, preselected offsets in 1/100th frame increments.

When the Shadow is used as a controller, the engineer can choose one of three different modes to determine the tightness of the lock between master and slave. FRAME LOCK locks both machines by using all the information contained within the time code format, providing a lock so tight that even the master's speed deviations are passed onto the slave. AUTO LOCK first provides Frame Lock-type synchronization until the transports are within one subframe (1/100th frame) of each other; the Shadow System then switches to SYNC LOCK. SYNC LOCK gives a lock that strictly resolves the time code information from the slave to the master input reference without regard to the absolute value of the displayed code. SYNC LOCK also provides a "software filter" which virtually eliminates wow and flutter transfer from master to slave. This is achieved by calculating a highly-accurate average tape speed, referenced to the master. Then, the Shadow matches the slave machine to that tape speed.

When operating the Shadow as a synchronizing auto-locator, the engineer can pick the edit point, and then simply enter the numbers representing the frame location into the Command Console keyboard or RS.232C terminal display. He would then hit one more button—GO TO—and both master and slave transports will proceed to that edit point. The engineer may then choose to do edits with subframe-accurate record-ins, record-outs, or mutes, which are all programmable from the Command Console or the RS.232C port. (Note that the Shadow Command Console was designed as an *alternative* to the use of the RS.232C communications port.)

By using its internal switching system, the Shadow can synchronize to standard time code (SMPTE or EBU, drop frame or regular), 24-frame film code, 60-Hertz auxiliary reference, or a wide variety of external video inputs. These sync options are especially convenient when code must be laid down during production in the field.

The ability to work with different codes is provided by an internal comparison method that allows the engineer to synchronize any two types of similar 60 Hz waveforms. A slave machine with time code may be synched to any of these master sources: time code, vertical drive, composite sync, or 60 Hz reference. It is also possible to synchronize any two machines that don't have time code, as long as they both supply 60 Hz pulses. (The Shadow, however, operates to its fullest potential when working with time code.) The Shadow can also synchronize bad or damaged tape, or tape with faulty or deteriorated time code. This is accomplished by referring to the tach pulses generated by the tape machine's control track head or tach idler. This feature enables the Shadow to automatically shift between time code and tach pulses whenever the time code degenerates or stops. The Shadow refers to the pulses to update the normal time code display until good code begins again (tach pulses are not used for synchronization).

During power-up, the Shadow performs a series of self-diagnostic tests, allowing the engineer to check performance level. Error codes are displayed on an LED read-out. To effectively interlock one machine to another, the engineer need only press the CHASE ENABLE button on the Shadow's front panel. The Shadow then works independently, providing exact and swift synchronization.

All interfaces to the master and slave machines are made through the Shadow. Compatible with most audio, video, and film transports, the Shadow's control loop enables it to automatically adapt to each required application, so switching from one process to another is a quick and easy process.

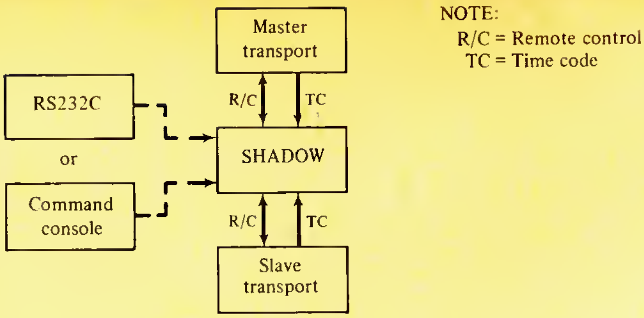


Figure 3. A two-machine, chasing slave application.

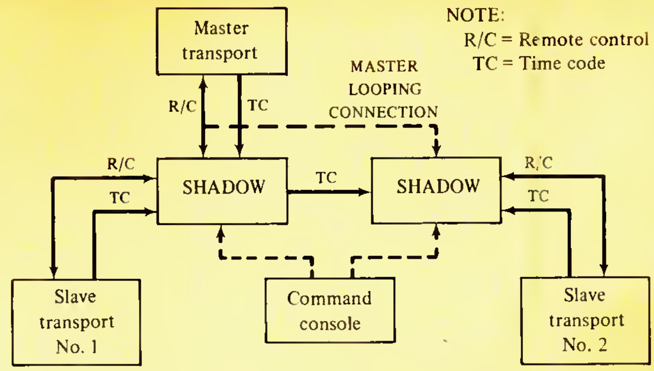


Figure 4. A two chasing slave machine application.

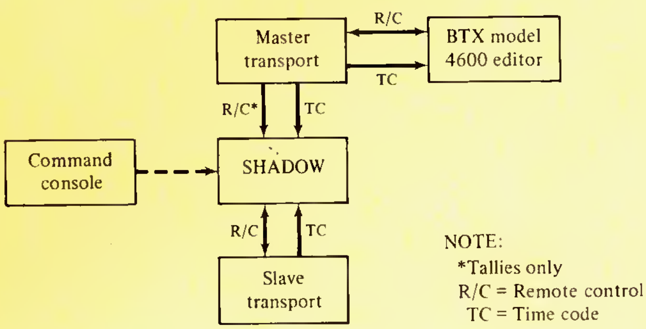


Figure 5. One editing master, one chasing slave.

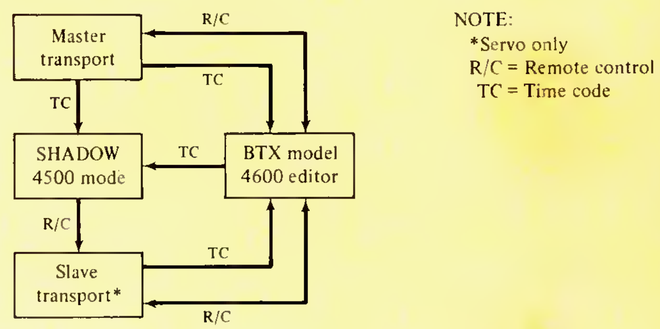


Figure 6. A two-machine editing, locking application (no chase).

INTERFACING THE SHADOW

The BTX Corporation provides the information needed (pin-out charts, schematics, etc.) to interface a particular brand/model tape machine with the Shadow. The tape machine being considered as a Shadow interface, as a master *or* slave, must have the following controls or indicators:

1. *transport speed-indicating signal* (i.e., tach pulses or control track pulses). This signal must come at a rate of over four per second at normal play speed, and less than 2.5 kHz at the fastest wind speed.
2. *transport direction signal*, which indicates the true direction of the transport, and can be of either logic convention. The Shadow automatically learns which convention is correct.
3. *play, stop, rewind, fast forward tallies* or *lamps*.
4. *tally supply* (reference supply for the tallies).
5. *command supply* (reference supply for all control functions coming from the Shadow).
6. *play, stop, rewind, fast forward commands*.

For a tape machine to be used as a slave transport in a Shadow application, it must also have a *speed over-ride* control (either a frequency—15 Hz to 40 kHz—or a voltage control, in a variable, calibratable range from half normal play speed to twice play speed). Although less range is acceptable, the lock-up time will be proportionally longer.

The Shadow is designed to interface one master transport to one slave transport. To synchronize three or more transports, two or more Shadows are necessary. A special looping connection cable parallels all the master transport control connections of each Shadow to be used.

Electronic synchronization offers unlimited prospects for the future of the audio industry. As record production becomes more complicated, as the video-disk climbs in popularity, and as the need for additional tracks rises, digital synchronization of multi-track machines becomes truly invaluable.

In addition, there is powerful potential for automation systems using interfaces such as the Shadow, which provide the audio studio with the high-tech editing and control capabilities now commonplace in the video industry.

Even with the availability of low-cost electronic synchronizers, there are still a lot of razor cuts being made in the editing room. It is now left to studio management to retire the razor blade by taking full advantage of computerized control systems. ■

JACK K. GORDON

Monitoring Program Levels

The Program Level Meter has been taken for granted for so long that the multiple role it must play is in danger of being forgotten.

THE RELATIVE MERITS of the VU Meter and the PPM have been carefully weighed during the past few years, and excellent analyses are available, such as those by Schmid (1, 2) and Silver (3). Further direct comparisons are unlikely to reveal much that is new. In fact, in the past forty years, the only changes to the VU meter have been cosmetic. And while early use of the PPM was largely restricted to government-funded organizations like the BBC (until transistors eliminated the bulk and cost of the tubes hidden behind it), the PPM has changed even less. In all essentials, each is still the same instrument that was around in the days of the steam locomotive. Each is an intact example of late-1930s audio technology.

If a reasonably well-informed stranger were to attend a seminar on program-level meters, he might well note something incongruous about it. The discussion, if it followed the usual pattern, would progress something like this: First, the advanced technology of the latest luminescent displays would receive its due consideration. Then discussion would edge away, to focus instead on the relative merits of the VU meter and the PPM. The discussion might well go further, to become a serious debate as to whether or not one of these forty-year-old instruments (the PPM) should take precedence over the other forty-year-old instrument (the VU Meter) as today's recognized standard for measuring program-levels.

By this time, our hypothetical observer would be having a hard time keeping a straight face. He had come for a first-hand glimpse of advanced audio technology, but what he had found instead was steam-age indicators still being used to monitor all that space-age consoles do. This is the preposterous picture that emerges when familiar facts are rearranged. They have been placed in this perspective to make one point, that comparing the VU Meter with the PPM does not explain the remarkable fact that both are still around to be compared.

Anything in audio that remains changeless for decades is certainly remarkable, but it is the nature of this dual survival that is of particular interest. The VU meter and the PPM are dissimilar instruments. Yet together they rapidly displaced all other devices, ended a period of minor chaos in program-level monitoring, and have peacefully coexisted, year after year, until the use of one or the other came to be taken for granted. In no way is this a history of conflict between opposing concepts of what a program-level meter should be. If the two concepts had been in conflict, one instrument would have displaced the other long ago.

Jack K. Gordon is a retired audio and broadcast design consultant. He is a member of the AES and a senior member of the SBE.

Concerns beyond those of measurement were certainly considered in the designs of the instruments, but these concerns were not set down. For to be in audio at that time was to be involved in its every aspect. These concerns were common knowledge, to be passed on as a matter of course. But times have changed. And although the instruments haven't changed in forty years, the people who use them have.

As Les Paul put it during an interview (6), the engineer who works the top of the board and the one who masters the complexities below it are no longer interchangeable. The program-level meter sits precisely at the interface between them. It is a common reference-point, but it is not always seen by both in quite the same light. And so, to discuss technical characteristics alone, is to lose sight of those other factors that are as important today as they ever were.

Measurement concerns and concerns beyond measurement are parallel aspects of program-level monitoring. It was full recognition of their interdependence that led to the two complementary designs which proved so successful. At a time of change, it becomes prudent to carefully re-examine all aspects in such a way that options can be seen by all to meet the needs of all. That is the objective of the discussion which follows.

The characteristics of the two instruments were set down in the form of specifications (4, 5), and certainly the technical differences between the average response of one and the quasi-peak response of the other suggest constrasting operational design-concepts. But if the two meters are compared in terms of their "unread peak" characteristics, how they operate can be disregarded. This makes it much easier to see how they have co-existed for so long without conflict, and to evaluate the degree of conflict that finally has arisen.

MEASUREMENT

Highs and lows disappear at the microphone. At that point, spectrum information is encoded in the single dimension of voltage amplitude. Whether the encoding is analog or digital, it is only when the information is decoded by a loudspeaker that highs and lows, independent sound-waves in three-dimensional space, reappear. Within the single dimension of voltage, no highs or lows exist. Such terms as "boosting highs" describe the end result, which is attained by manipulating the encoding.

What goes on in that single dimension is almost embarrassingly simple. When sound-waves flow past a microphone, the voltage it generates responds to those waves exactly as a cork responds to the waves on a rippling pond. It is hardly news that the bobbing cork only knows the single dimension of up-and-down. To see that single dimension, first display the waveform (a visual decoding) on an oscilloscope, then kill the sweep. The vertical line that remains is the reality of the single dimension, and its vertical bouncing is the reality of the encoding.

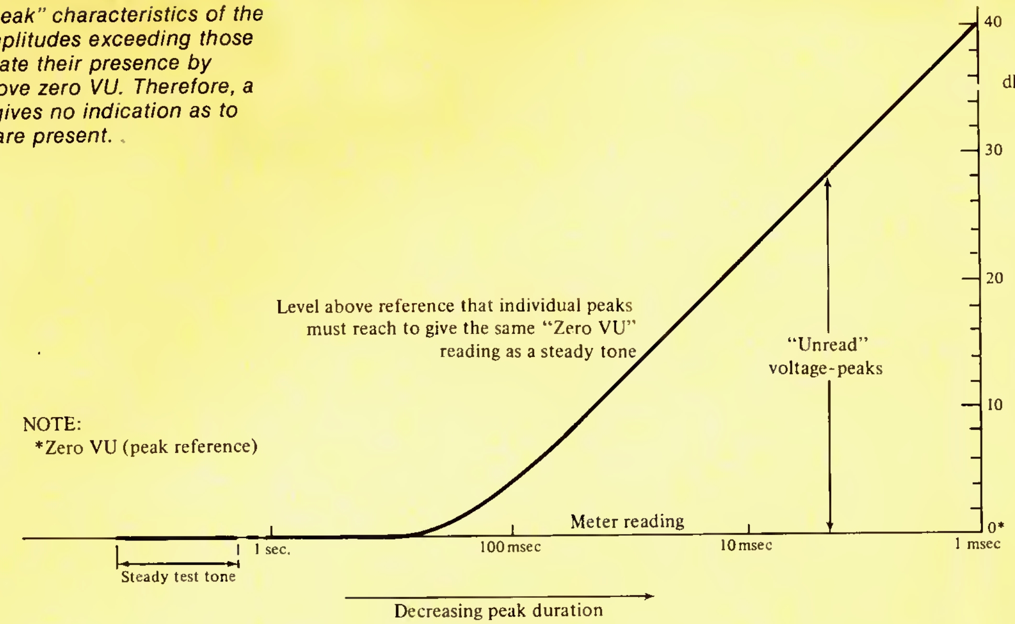


Figure 1. The basic "unread peak" characteristics of the VU meter. Only peaks with amplitudes exceeding those defined by the curve will indicate their presence by producing a meter reading above zero VU. Therefore, a zero VU reading on program gives no indication as to whether or not unread peaks are present.

The relation between voltage peaks and program peaks or program content (sound) is therefore far from direct. The random factor is evident in the way that a cork may be lifted by a single wavecrest, or shot skyward by the chance simultaneous arrival of many smaller wavecrests from many sources. It is the chance simultaneous arrival of wavecrests at the microphone that results in the highest and fastest voltage-peaks at its output, not the sound itself. Random strings of such high-amplitude voltage-peaks constitute the "peaks" which a program-level meter must sort out and read. The probability that they will be reproduced by the meter increases rapidly with the density of the sound spectrum, but otherwise the relation is unpredictable.

The sky may be the limit for the bouncing cork, but every amplifier has a ceiling above which the voltage cannot rise. If it tries to exceed the available headroom, it will clip. When this happens, bits of the original encoding will be lost, and replaced by false bits. The false bits will be decoded by the loudspeaker to produce alien sound-waves. Some will be heard as distortion. Others will not.

At BBC Engineering some forty years ago (7), psycho-acoustic research into this aspect of hearing established the following: If a peak (or a string of voltage peaks) lasts longer than 10 milliseconds, the alien sound of clipping will be heard as distortion. If the duration of the clipped peak is less than 10 milliseconds, alien sound will still be reproduced by the loud-speaker, but it will not be heard.

Why this should be so is reasonably well-understood. Sounds to which the tone-deaf eardrum responds are recreated as true sound-waves in the fluids of the inner ear. There, a membrane floats at the interface between two fluids, and like a ribbon floating on a pond, it conforms to the changing *shapes* of the passing waves. (No single dimension here.) Some twenty thousand nerve cells along its length convey wave-shape information to the brain. But the cranial computer requires a split-second to identify the individual sound-waves present in two inner ears, and to produce a mental picture of the sound sources and their locations in three-dimensional space. In ways unknown, sounds too brief to be properly identified are ignored. They are simply not heard.

Years of experience have confirmed the BBC finding that in the case of the alien sounds created as the result of *clean* clipping, 10 milliseconds divides those which are heard from those which are not. (More on this later.) The first measurement function of a program-level meter is therefore to warn of peaks which will result in audible distortion if clipped. The second is to ignore peaks which may be clipped with impunity. It is this second function that permits a program-level meter to be evaluated in terms of its "unread peak" characteristic. Unread peaks are those that have amplitudes above the peak reference-level, but which are too brief for the meter to respond with a reading that indicates their presence.

The shorter the duration of an unread peak, the higher its level must be to drive a meter to the peak-reference mark. The curve of FIGURE 1 is a plot of the increasing level that peaks shorter than 300 milliseconds must attain to drive a VU meter to a zero reading. It can be seen that peaks, generated at random by program material, may momentarily attain very high amplitudes without exceeding the limits represented by the curve. Any may be present at any amplitude up to that limit, while the meter continues to peak no higher than zero. These are the unread peaks that the VU meter ignores.

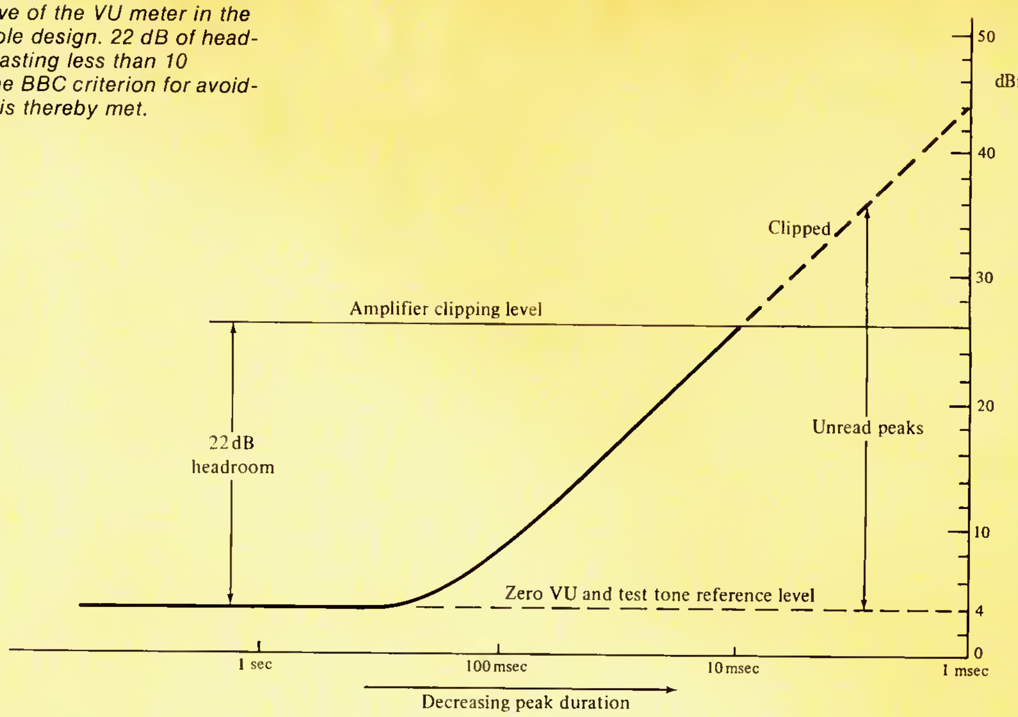
In FIGURE 2, the unread-peak curve has been placed in the context of modern console design. The test-tone reference-level for a reading of zero VU is +4 dBm and the clipping-level of the output amplifier is +26 dBm, fairly typical of today's practice. It is presumed that the program level is not allowed to exceed zero VU. Under these conditions, only unread peaks lasting less than 10 milliseconds will be clipped. Unread peaks of greater duration may occupy available headroom, but none will be clipped. Clearly, the 10 millisecond criterion is fully met by such consoles, and has been for many years.

In FIGURE 3, the clipping level is presumed to be +12 dBm, as might be the case when feeding a tape-recorder. It is representative of situations where additional headroom cannot be provided without raising the noise-level. Under these conditions, unread peaks lasting as long as 60 milliseconds may be clipped. The shaded area indicates the duration and amplitude of unread peaks which, if present, will be clipped to produce audible distortion.

At the time the VU meter was introduced, random unread peaks in excess of 6 dB above reference were so rare that the risk of audible distortion was minimal. The risk remains relatively low when monitoring pre-processed material, and in general, 6 dB can still be taken as the amplitude above reference of "average" unread peaks. But disc-reproduction has begun to generate peaks of significantly higher amplitude, and modern wide-range microphones, along with close miking and multiple miking, have made the probability of audible distortion unacceptably high.

The BBC opted for the no-risk route from the beginning, bulky electronics notwithstanding. In FIGURE 4, the unread-peak curve of the PPM has been placed in the same tape-recorder context as the VU meter curve of FIGURE 3. The test-tone reference-level remains +4 dBm, which corresponds to a

Figure 2. The unread-peak curve of the VU meter in the context of contemporary console design. 22 dB of headroom assures that only peaks lasting less than 10 milliseconds will be clipped. The BBC criterion for avoiding audible clipping distortion is thereby met.



down-scale mark on the standard PPM. The peak-level reference mark (equivalent to the zero of the VU meter scale) corresponds to a steady-tone level of + 10 dBm, 6 dB above the test-tone level. That additional 6 dB accommodates "average" peaks. It can be seen that only unread peaks lasting less than 10 milliseconds will be allowed to clip. Peaks of longer duration are directly and accurately read, and can safely be held within 2 dB of clipping, with assurance of a positive meter-indication if any exceed that level.

When calibrated as shown in FIGURES 3 and 4, with a differential between the two peak reference-levels of 6 dB, the meters will tend to peak in the same range on material having average peak-levels. They will differ only when the PPM detects what the VU meter does not. The standard differential of 6 dB is empirical. Schmid (2) has suggested that a better average match can be obtained with a differential of 8 dB.

Despite the fact that a simulated agreement in readings can be obtained by this means, the unread-peak characteristic of the PPM is clearly superior to that of the VU meter in avoiding audible clipping-distortion. The need for its extra protection can only increase with time. On these specific points there is no room for argument.

BEYOND MEASUREMENT

Two aspects of program-level display will be examined. The first is communication efficiency. The second is display behavior.

The pivoted pointer, used in both the VU meter and the PPM display, has two unique properties that contribute significantly to easy and effective visual communication. The first, deceptively simple, is angle. The powerful effectiveness of angle is evident whenever we glance at the hands of a clock. There's no need to read the numbers and no need to match anything to a scale. Two pointers convey the time by means of angle alone. In the same way, the angle of the pointer in a program-level meter instantly conveys what needs to be known. The actual scale is rarely noticed during normal operation.

The second property is fan-of-motion. Any rally-driver, who must avoid red-lining his tachometer as he busily downshifts through hairpin bends, can attest to the superiority of the pivoted pointer to any other active display. The tip of a pointer may move so fast that it becomes invisible, but velocity is not uniform along its length, and its angular sweep creates a visible fan-of-motion. Peripheral vision is particularly sensitive to motion of this kind, and a fan-of-motion can send a clear signal to the eyes even when they are focused elsewhere. In today's busy control-rooms, assured peripheral communication is an important asset.

These contributions to efficient communiction were carefully weighed when galvanometers were chosen over the flashing lights of the day for VU meter and PPM displays. The importance of the pivoted pointer was ultimately condensed into a single phrase that appears in the British specification for

Figure 3. The unread-peak curve of the VU meter in the context of feeding a tape recorder, which is assumed to clip at a level 8 dB above the test-tone level. Unread peaks lasting as long as 60 milliseconds may be clipped. The hatched area represents the amplitude and the duration of unread peaks which, if present, will clip to cause audible distortion.

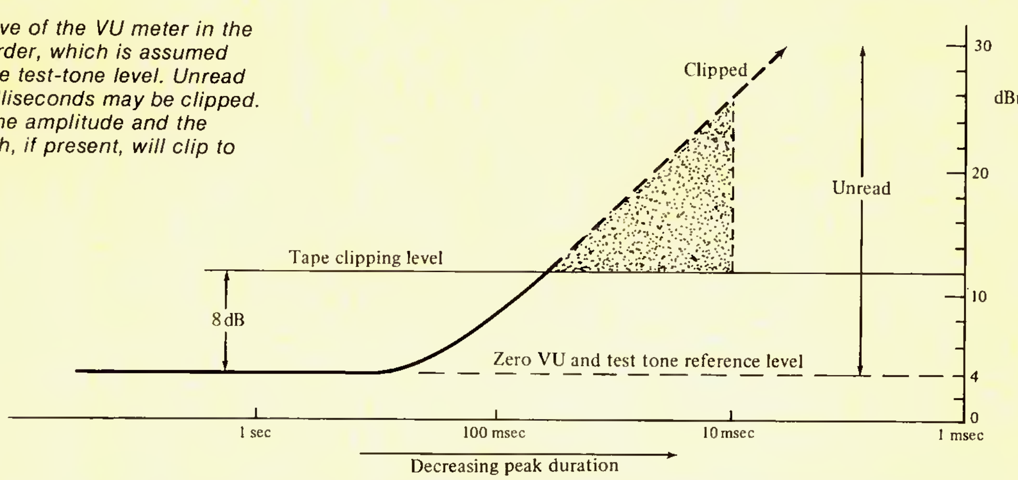
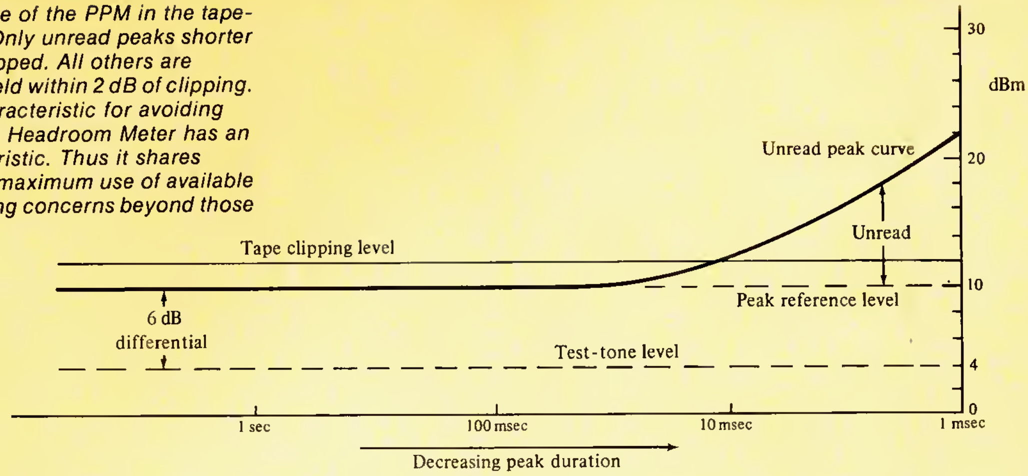


Figure 4. The unread peak curve of the PPM in the tape-recording context of Figure 3. Only unread peaks shorter than 10 milliseconds will be clipped. All others are indicated directly, and can be held within 2 dB of clipping. This is the BBC-developed characteristic for avoiding audible clipping distortion. The Headroom Meter has an identical unread-peak characteristic. Thus it shares the ability of the PPM to make maximum use of available headroom, while accommodating concerns beyond those of measurement alone.



the PPM. It says that "...not less than 75 percent of its length shall be clearly visible at any reading." The details are missing, but the impliction is clear. A fully visible pointer contributes uniquely to easy and effective communication. And, if all the characteristics of pointer and scale are chosen for a visual optimum, pointer communication can be very easy on the eyes.

There are two divergent views as to what information the display-behavior should convey. Both views deserve to be fully understood.

A network engineer expressed his view quite forcibly. "The VU meter is useless!" An understandable statement. For hours he must tastefully fit programs he has never heard before into one small slot in the telecommunications mosaic. Through the other slots pour unending streams of analog and digital data. For his program to arrive intact at its destination, it must override a background of unpredictable electronic yammer without peaks shattering at any point along a thousand-mile route of common-carrier equipment. Even those peaks that *may* be clipped, *must* be clipped. If they enter the network, they can splatter to play havoc in other channels.

In these circumstances, nothing quite equals the way that the pointer of the standard PPM flicks to an unequivocal peak-reading, and then glides smoothly downward until the next significant peak flicks it to a new reading. All that is needed for precision gain-riding is clearly displayed, with no clutter of pointer-activity between peak-readings to distract and fatigue. In a context such as this, the PPM is an ideal instrument. It is not at all unreasonable that the VU meter should be considered useless by comparison.

The PPM originated in Europe, at a time when broadcasting dominated audio, and where state-operated networks such as the BBC held the exclusive right to broadcast. The PPM, ideal for the network engineer, naturally rose to dominance. The VU meter was conceived in North America, where in addition to networks large and small, thousands of private stations originated live broadcast of their own. Emphasis was on the needs of innumerable teams and individuals constantly involved in the pickup and direct broadcast of live music. Not surprisingly, reluctance to abandon the VU meter is primarily encountered in those areas of audio that involve live performances.

When asked why he continued to use the VU meter even though PPMs were available to him, a respected mixer-musician paused to reflect, and then said, "The VU meter talks to me. The PPM does not." His words were apt. They describe what the VU meter, originally the "NAB Standard Volume Level Indicator" (7), was designed to do. The objective selected for its display behavior was "syllabic response." Its designers reasoned that if a meter were to faithfully follow the rhythmic cadences of the syllables of human speech, then what is seen when monitoring a vocal performance would be a plausible dynamic replica of what is heard. It was further reasoned that a meter with syllabic response would track the complexities of musical cadences closely enough that good eye/ear rapport would also be attained when monitoring musical performances.

Instant and relatively permanent acceptance confirmed the wisdom of choosing this design objective. The instrument that achieved it in conjunction with an acceptable "unread peak" characteristic was deceptively simple. As the years grew into decades, the self-evident "rightness" of VU meter display behavior came to be taken for granted. The skillful engineering behind that behavior was gradually forgotten. When the time finally came for change, its measurement characteristics were still in evidence to be examined and discussed, but its design-objective was not. A reassessment of "eye/ear rapport" in today's context is therefore in order.

The art of the studio engineer is to create the illusion of reality. Not reality itself, for that is impossible. A loudspeaker may simulate a single instrument very well, but no matter how sophisticated loudspeakers may become, vibrating diaphragms in a pair of wooden boxes can never actually be a real orchestra in a real concert hall, or even really be a small combo. Sound-pressure variations at the loudspeaker which duplicate those sensed by the microphone provide an ideal starting-point, and designing equipment to achieve this is an art in itself. But this step only re-creates the arithmetic of music.

It is true that the arithmetic alone can clearly communicate to others musically adept, but only the performers and their performance will communicate to the majority of listeners. The ultimate art of audio is to skillfully and intuitively manipulate electronic encoding until a pair of loudspeakers create the *illusion* that performers are present. The modified encoding that achieves the illusion is what must be captured and preserved. Today, the illusion is often entirely synthetic, created from bits and pieces that were never together as a reality. So superb "reproduction" is in fact not reproduction at all. It is a skillfully created illusion.

Every subtle aid to the creation of that illusion is important. If mood lighting in a control-room assists, it is important. This is equally true for the mixer, the musician, and for the disc-jockey who strives to weave words and recordings into a flowing continuum of sound. But those who value an atmosphere of total rapport within the creative environment face a dilemma. If the VU meter has become technically inadequate, it must go. If that means the loss of eye/ear rapport, it must stay. It is a no-win situation. Either choice means a loss to individuals, and hence a loss to audio as a whole.

If all needs are to be met without conflict or loss, a replacement for the VU meter is needed which complements the PPM. Its unread-peak characteristic should match that of the well established PPM without compromise. Its interpretation of musical dynamics should be at least equal to that provided by the VU meter. And if at all possible, means should be found to obtain comparable performance from those existing VU meters which, for practical reasons, cannot readily be replaced. These are the design-objectives which will now be discussed.

EVOLVING A SPECIFICATION

The strength of the VU meter design lay in the choice of a real-life model for its display-behavior. Syllabic cadence could be timed, and timing could be reduced to a mean, to a number (300 milliseconds). Without a real-life model, modifying display-behavior is guesswork. Syllabic response leads inexorably to the original VU meter design, and to its uniquely inter-dependent characteristics of display and measurement. So an alternative real-life model for "artistic" display-behavior had to be found that would be compatibile with the unread-peak characteristics of the PPM.

A few years ago, idle curiosity as to what sort of model might serve led to an increasingly wide-ranging series of observations and tests. The details of that initial investigation have been described elsewhere (8). Its conclusions can be briefly summarized.

The model finally chosen for display-behavior was the baton with which the symphony conductor visually conveys to his orchestra the cadence and the sound-intensity that he wishes it to produce. If the "baton" in a program-level meter conveys cadence and sound-intensity in the same manner, a high degree of rapport can be expected between what is seen and what is heard. Since no pointer can duplicate all the body-language used by a conductor, a simplified model was derived, based upon the following five elements of the conductor's art:

(1) A decisive upbeat which signals release of sound at the intensity indicated by the speed and the height of upward sweep; (2) A downbeat that arcs into an accelerating drop that is as decisive as the upbeat; (3) A fleeting but authoritative dwell at the crest of major upbeats that may be extended to indicate sustained sound; (4) Increased baton-activity at lower levels to emphasize subtle musical dynamics; (5) A "scale" for indicating sound-intensity which is variable, but which can be acceptably approximated by a characteristic midway between linear and logarithmic.

The fallback of the pointer in the standard PPM is controlled by an exponential capacitor-discharge curve. The curve is concave, resulting in a fallback speed that decreases as the pointer drops towards its resting position. It is this feature of the PPM which suppresses the "clutter of pointer-activity" at lower levels. In contrast, the baton-model calls for a pointer fall-back curve that is convex rather than concave, one that provides controlled downward acceleration rather than deceleration, and which results in pointer-activity that is emphasized at lower levels rather than suppressed. A completely new electronic design was therefore developed to integrate baton behavior with the unread-peak characteristic of the PPM.

Design details of a new "Headroom Meter" will be described in Part II. The name was chosen for two reasons. First, it distinguishes the meter from the PPM, while conveying that it too permits maximum use to be made of available headroom with minimum risk of distortion. Secondly, as discussed in Part III, the term "VU/HU" can be used to describe the conversion which permits comparable performance to be obtained from unmodified VU meters.

Finally, the design proved amenable to the incorporation of two useful alternative modes of operation to which single meters or banks of meters can be conveniently remote switched. Discussion of the alternative modes is included in Parts II and III.

The result of all this is a program-level meter which gives peak readings identical to those of the standard PPM, but which conveys the musical dynamics that contribute to creativity in a way that the PPM cannot. A spin-off is means to upgrade existing VU meters so that they can better meet the demands of modern program-level monitoring.

CLIPPING, SIGNAL-PROCESSING AND LOUDNESS

The most critical step in audio is to initially capture the highest possible sound quality. Thereafter the problem becomes one of preserving that quality during the ensuing stages through which the signal must pass before it reaches the listener (e.g. recording, transmission, reproduction, etc.). Quality must be preserved despite a steadily decreasing dynamic range. Signal processors play an important role in the process.

In discussing the Headroom meter and the PPM, emphasis has been placed on the value of their shared unread-peak characteristic in avoiding clipping distortion during initial capture. Its value is equally high when driving signal processors. If a VU meter is used, high-amplitude unread peaks can drive a signal processor crazy, while at the same setting the processor will be barely tickled if material contains only low-level peaks. The VU meter cannot distinguish one from the other, so there is no way it can keep the signal processor operating within its optimum range. Both the Headroom meter and the PPM, which accurately read all peaks but those which can be clipped with impunity, will keep the processor operating within its optimum range at all times. The improvement in overall quality can be significant.

If signal processing is carried to an extreme, an audio signal can become so heavily reverbed, clipped and squared that any peak-reading meter will flick to a reading and just sit there. A VU meter, which ignores peaks rather dramatically, will still indicate the varying loudness of material which has been compressed to fit within those tight peak-limits. But the designers of the VU meter were careful to use the term "volume" rather than "loudness." Loudness can also denote sound intensity. When dynamic range is less restricted, either the Headroom meter or the PPM will indicate the intensity of plucked or percussive sounds far more accurately than does the VU meter. Preserving such transient detail has become increasingly important at all stages of audio processing. ■

Author Gordon will conclude this discussion in the January issue of **db**.

REFERENCES:

1. H. Schmid, "Audio Program Level, the VU Meter, and the Peak-Program Meter," IEEE Transactions on Broadcasting, Vol. BC-23, No. 1, March 1977.
2. H. Schmid, "The Peak Program Meter and the VU Meter in Broadcasting," Presented at the 67th AES Convention, New York, NY, Oct. 31/Nov. 3, 1980: Preprint No. 1691.
3. S. L. Silver, "VU Meters vs. Peak Program Meters": **db** Magazine, Jan./80.
4. British Standard 4297:1968, "Specifications for the Characteristics and Performance of a Peak Programme Meter."
5. ASA Standard C16.5-1942, "American Recommended Practice for Volume Measurement of Electrical Speech and Program Waves": (Also ANSI C16.5-1954 and C16.5-1961).
6. P. Lawrence and R. Rypinski, "Interview with Les Paul": AUDIO, Dec./78, p. 60.
7. (NOTE: Descriptions of the developments that began in the late 1930s are based primarily on the clear memory of the author. Early psychoacoustic research at BBC Engineering has been attributed on good verbal authority to C. G. Mayo, and development of the VU meter to H. A. Chrinn, D. K. Gannet and Ron Morris. Not surprisingly, obtaining full documentation at this late date has proved impractical.)
8. J. K. Gordon and J. B. Wood, "Bridging the Gap between the VU Meter and the PPM": Presented at the 64th AES Convention, New York, NY, Nov. 2/Nov. 5, 1979: Preprint No. 1518.

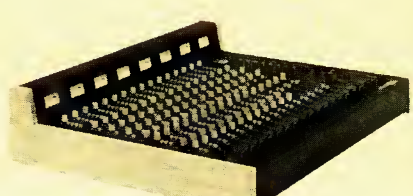
db New Products & Services

PORTABLE LOUDSPEAKER SYSTEM

• The Shure Model 711 two-way loudspeaker system was designed with the needs of small to medium performing groups in mind, but it is also suitable for sound reinforcement applications in schools, churches and auditoriums. Each speaker system consists of the new Shure-designed 15-inch, die-cast frame woofer in a bass reflex cabinet, and a Shure high-frequency horn and driver combination. Both woofer and horn are front-mounted. Power handling capacity is 150 watts of continuous program material. The 711 has an impedance of 8 ohms and produces 101 dB SPL at one meter with only a one-watt input. A unique feature of the system's horn is the VARAD variable sound dispersion control. Operation involves adjusting two sliding controls for one of four sound dispersion patterns: 60 degrees, 90 degrees left, 90 degrees right, or 120 degrees. The 60 degree setting is used where narrow, long-throw coverage is desired; the 90 degree settings are for medium range coverage, or for odd-shaped rooms, and the 120 degree setting is for wide area coverage in short-throw applications. The Model 711 loudspeakers are compatible with power consoles such as the Shure Models 700 and 706. Rear-mounted jacks on the speaker cabinet allow the option of external bi-amplification, without internal wiring changes.
Mfr: Shure Bros., Inc.
Price: $590.00
Circle 64 on Reader Service Card

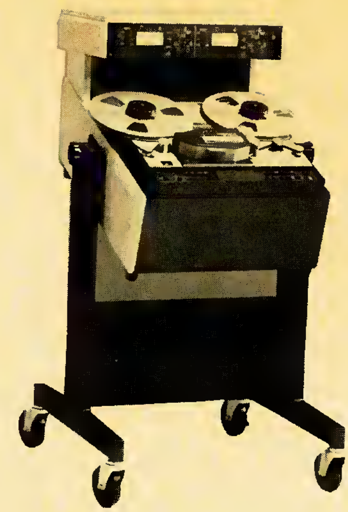
CONSOLE

• The Speckmix 16 is a 16 input, 8 output recording console reseached, designed, and intended specifically for professional and semi-professional 8 track studios. Features include: 16 complete input channels; 8 mixing buss outputs; 8 large V.U. 8 track panable assign; 3 band equalizers; low noise transformerless mike inputs; control room and studio playback, talkback and cue prompts; and independent stereo mixdown buss. All operating levels are ±4 dBm. Basic specifications include: frequency response 23 Hz-20 kHz (±1 dB), output level +4 dBm, maximum output level +22 dBv, noise −72 dB (mike input to buss output), and noise −80 (line input to program feed output).
Mfr: Speck Electronics
Price: $3,575.00
Circle 66 on Reader Service Card



AUDIO RECORDER

• The ATR-800 audio recorder is designed for the radio and television broadcaster and recording studio professional. A cue amplifier has been built into the system to permit monitoring of a single channel or multiple channels of the tape during editing or while it is being cued. The system also has hands-on-reel and tape dump modes for manual editing convenience and is equipped with a variable shuttle control to regulate speed and direction, making editing simpler. The electronic tape timer further enhances the editing capabilities of the recorder by providing accurate timing of recorded segments for all three speeds (7½, 15, 30 ips) in hours, minutes and seconds. Standard on the ATR-800 is the single point search-to-cue. With this feature, the recorder will automatically return to a designated cue or edit point. Also standard on the system is the Pick Up Record Capability (PURC) which allows editing or dubbing of new material without creating errors at either end of the new insert on the tape. The recorder has been designed to accommodate a fourth head for playback of two different tape formats without requiring a head assembly change. The variable speed operation on the ATR-800 permits recording and play-



back up to ±10 percent of nominal speed to correct recordings made off speed or for special effects. The tape marker and cutter options designed to mark and cut edits is available for the first time from Ampex on the ATR-800. The ATR-800 is available in one, two, and four channel configurations.
Mfr: Ampex Corporation
Price: $5,450.00
Circle 67 on Reader Service Card

db Classified

Closing date is the fifteenth of the second month preceding the date of issue.
Send copies to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road, Plainview, New York 11803

Minimum order accepted: $25.00
Rates: $1.00 a word
Boxed Ads: $40.00 per column inch
db Box Number: $8.50 for wording "Dept. XX," etc.
Plus $1.50 to cover postage

Frequency Discounts: 6 times, 15%; 12 times, 30%

ALL CLASSIFIED ADS MUST BE PREPAID.

FOR SALE

MICROPHONES BY UPS. Quicker. You'll save more with us. All popular models for immediate delivery. **UAR Professional Systems. (512) 690-8888.**

JFET TUBE REPLACEMENTS for first playback stages in most Ampex Professional audio tape recorders/reproducers available from **VIF International, Box 1555, Mountain View, CA 94042. (408) 739-9740.**

AKG BX20 reverberation and C414 microphones. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

AMPEX MM-1200 16 track with VS-10 Varispeed. Like new, less than 2,000 hours. If purchased new $29,200, sacrifice $20,000. **(713) 781-8281.**

SCULLY 280-FT MONO RECORDER
New Nortronics Duracore heads.
Tentrol constant tension.
Many key parts recently replaced.
Ampex 440-C console cabinet.
by owner—**Rob Edward**
(914) 946-4244

REELS AND BOXES 5" and 7" large and small hubs, heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116 (214) 296-2773.**

TASCAM MODEL 5, like new, full warranty, $1,100.00. Two Electro-Voice RE-15's, $150.00 each, RE-20's, $300.00 each. **N.A.B., Box 7, Ottawa, IL 61350.**

TECHNICS, BGW, EVENTIDE, AKG, Scully, and many more IN STOCK. FOR IMMEDIATE DELIVERY: **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. (512) 690-8888.**

PROFESSIONAL AUDIO EQUIPMENT
Shop for pro audio from N.Y.'s leader, no matter where you live! Use the Harvey Pro Hot-Line. (800) 223-2642 (except NY, AK, & HI). Expert advice, broadest selection such as: Otari, EXR, Ampex, Tascam and more. Write or call for price or product info:
Harvey Professional Products Division
2 W. 45th Street
New York, NY 10036
(212) 921-5920

LEXICON 224 Digital Reverberation. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229, 512-690-8888.**

AMPEX, OTARI, SCULLY—In stock, all major professional lines; top dollar trade-ins; write or call for prices. **Professional Audio Video Corporation, 384 Grand Street, Paterson, New Jersey 07505. (201) 523-3333.**

THE LIBRARY...Sound effects recorded in STEREO using Dolby throughout. Over 350 effects on ten discs. $100.00. Write **The Library, P.O. Box 18145, Denver, Colo. 80218.**

ORBAN. All products in stock. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

FREE CATALOG & AUDIO APPLICATIONS

CONSOLES
KITS & WIRED
AMPLIFIERS
MIC., EQ, ACN, LINE, TAPE, DISC, POWER
OSCILLATORS
AUDIO, TAPE BIAS
POWER SUPPLIES
OPAMP LABS INC.
1033 N. SYCAMORE AVE.
LOS ANGELES, CA. 90038
(213) 934-3566

BLANK AUDIO AND VIDEO CASSETTES direct from manufacturer, below wholesale, any length cassettes: 4 different qualities to choose from. Bulk and reel mastertape—from ¼-inch to 2-inch. Cassette duplication also available. Brochure. **Andol Audio Products, Inc., Dept. db, 42-12 14th Ave., Brooklyn, NY 11219. Toll free 1-800-221-6578, ext. 1, NY residents (212) 435-7322.**

USED/MINT CONDITION: Fluke 8050A DMM, Comm Corp RTA ARA-412B + BONG-2, B&K #1801 40MHz frequency counter. **Carl Austin, Box 14225, St. Petersburg, Florida 33733.**

AMPEX, OTARI & SCULLY recorders in stock for immediate delivery; new and rebuilt, **RCI, 8550 2nd Ave., Silver Springs, MD 20910.** Write for complete product list.

LEXICON, dbx, & UREI. Most items for immediate delivery. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. (512) 690-8888.**

USED RECORDING equipment for sale. **Dan (415) 441-8934.**

SERVICES

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAECO, Grampian. Modifications done on Westrex. Quick turnaround. New and used cutterheads for sale. Send for free brochure: **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

ACOUSTIC CONSULTATION—Specializing in studios, control rooms, discos. Qualified personnel, reasonable rates. Acoustilog, Bruel & Kjaer, HP, Tektronix, Ivie, equipment calibrated on premises. Reverberation timer and RTA rentals. **Acoustilog, 19 Mercer Street, New York, NY 10013 (212) 925-1365.**

JBL AND GAUSS SPEAKER WARRANTY CENTER

Fast emergency service. Speaker reconing and repair. Compression driver diaphragms for immediate shipment. **NEWCOME SOUND, 4684 Indianola Avenue, Columbus, OH 43214. (614) 268-5605.**

VIF INTERNATIONAL will remanufacture your Ampex or Scully (Ashland/Bodine) direct drive capstan motor for $200. Average turn around time—2-3 weeks. For details write **PO Box 1555, Mtn. View, CA 94042, or phone (408) 739-9740.**

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. **IEM, 350 N. Eric Drive. Palatine, IL 60067. (312) 358-4622.**

EMPLOYMENT

ENGINEER/ENGINEER ASSISTANT seeking full time permanent work. 4 years audio, 2 years video, and maintenance experience. Equipment to offer. **Call (205) 263-6353.**

MAINTENANCE ENGINEER position open in top NYC recording studio. **Dept. 122, db Magazine, 1120 Old Country Rd., Plainview, NY 11803**

SEEKING ENTRY LEVEL position in audio recording studio or possibly hardware manufacturing. Graduating May '82, BSEE from Penn State U. Music background, good ear, some equipment repair. **Doug Arnold, 801 D-16 W. Aaron, State College, PA 16801, (814) 237-2713.**

BUSINESS OPPORTUNITIES

RECORDING STUDIO fully equipped, attractive, prime operating condition, Philadelphia area. **Dept. 120, db Magazine, 1120 Old Country Rd., Plainview, NY 11803.**

WANTED

WANTED: TRANSCRIPTION discs, any size, speed. Radio shows, music. **P.O. BOX 724—db, Redmond, WA 98052.**

Copies of db

Copies of all issues of db—The Sound Engineering Magazine starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to:

University Microfilm, Inc.
300 North Zeeb Road
Ann Arbor, Michigan 48106

Special binders now available.

All you regular **db** readers who, smartly enough, keep all your back issues, can now get our special binders to hold a whole year's worth of **db** magazines in neat order. No more torn-off covers, loose pages, mixed-up sequence. Twelve copies, January to December, can be maintained in proper order and good condition, so you can easily refer to any issue you need, any time, with no trouble.

They look great, too!

Made of fine quality royal blue vinyl, with a clear plastic pocket on the spine for indexing information, they make a handsome looking addition to your professional bookshelf.

Just $7.95 each, available in North America only. (Payable in U.S. currency drawn on U.S. banks.)

Sagamore Publishing Co., Inc.
1120 Old Country Road
Plainview, NY 11803

YES! Please send ______ **db** binders @ $7.95 each, plus applicable sales tax. Total amount enclosed $______.

Name______

Company______

Address______

City______

State/Zip______

Around the world, the choice of professionals is the

AUDIO SIGNAL DELAY.

From Westminster Abbey to performing arts centers such as Wolf Trap Park and Ravinia Festival, the Superdome and the Contemporary Hotel, Disney World, the digital AUDIO SIGNAL DELAY from Industrial Research Products has meant *PERFORMANCE* in these demanding facilities and many more!

ANNOUNCING . . . DF-4015

The NEW DF-4015 Digital Delay has the performance, reliability and flexibility that made the Audio Signal Delay the standard since the early 1970's, and is the most versatile model yet with even lower noise and distortion.

Check these outstanding features:

- 1, 2, 3 or 4 outputs
- 192 ms in 3 ms increments
- Thumbwheel delay setting
- Greater than 90 db dynamic range
- Compact 1¾" x 19" rack mount chassis
- Security panel
- Transformer isolated in and out

Model DF-4015

For more information,
CALL OR WRITE TODAY

INDUSTRIAL
RESEARCH
PRODUCTS, INC.

A *Knowles* COMPANY

321 Bond Street
Elk Grove Village, IL 60007
312-439-3600

Circle 11 on Reader Service Card

The New UREI Power Amplifiers

The UREI power amplifiers are designed to extend UREI quality from our low level signal processing all the way through to our exclusive Time Align™* studio monitors.

Careful evaluation of competitive power amplifiers indicates that while in some cases adequate reliability has been achieved, audible performance is often marginal.

The UREI 6000 Series amplifiers offer the knowledgeable professional the reliability he demands *and* the sound quality he deserves.

Exceptionally large power supplies and exceedingly high speed, high current circuitry combine to produce low-end punch and high-end sparkle which is unexcelled in professional power amps. Moderate use of negative feedback results in fast recovery from transient overloads and avoids excessive high order harmonic distortion when the amplifier is driven into clipping.

The Model 6500
Listening Amplifier

Two totally independent plug-in channels, removable from the front panel, each with its own power supply and continuously variable cooling fan. Exclusive Conductor Compensation™ corrects for wire loss and transducer related load anomalies, resulting in absolutely accurate waveforms at the speaker terminals. 275 Watts per channel into 8 ohms, 600 Watts per channel into 2 ohms. Standard rack mount, 7″ high.

The Model 6300
Dual Channel Power Amplifier

225 Watts per channel into 8 ohms, 380 Watts per channel into 4 ohms. 5¼″ rack space!

The Model 6250
Dual Channel Power Amplifier

150 Watts per channel into 8 ohms, 200 Watts per channel into 4 ohms. 3½″ rack space!

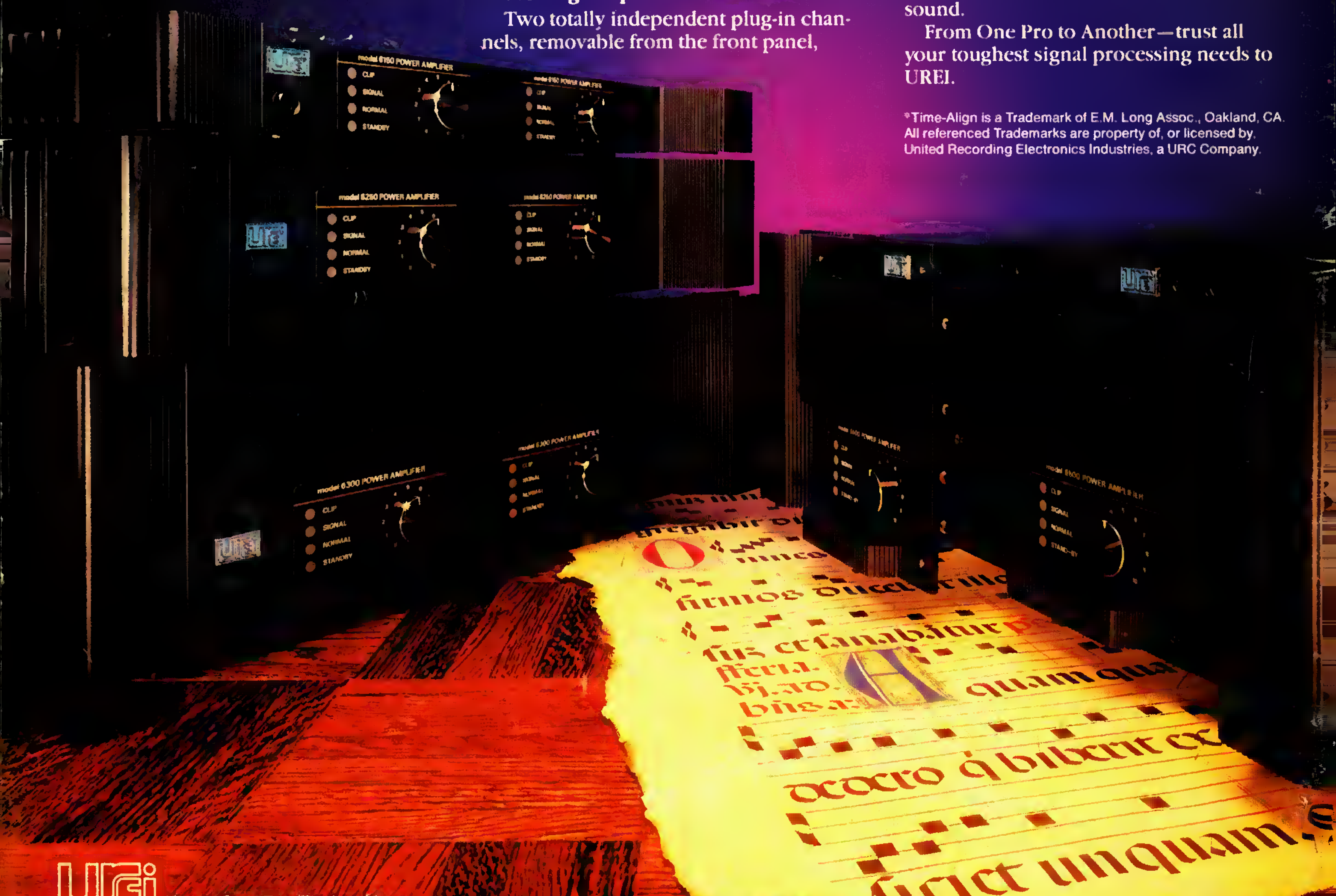
The Model 6150
Dual Channel Power Amplifier

80 Watts per channel into 8 ohms, 80 Watts per channel into 4 ohms. 1¾″ rack space!!

Audition the UREI Power Amplifiers at your professional sound dealer and discover how good a reliable amplifier can sound.

From One Pro to Another—trust all your toughest signal processing needs to UREI.

*Time-Align is a Trademark of E.M. Long Assoc., Oakland, CA. All referenced Trademarks are property of, or licensed by, United Recording Electronics Industries, a URC Company.



UREI From One Pro To Another
United Recording Electronics Industries
8460 San Fernando Road, Sun Valley, California 91352 (213) 767-1000 Telex: 65-1389 UREI SNVY
Worldwide: Gotham Export Corporation, New York; Canada: Gould Marketing, Montreal H4T 1E5

See your professional audio products dealer for full technical information.